



SHURE GUIDE To PERSONAL MONITORS

BY GINO SIGISMONDI



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We live in an age of extreme technological advances, and the audio industry is no exception. Average listeners, as well as musicians, have been conditioned by pristine, well-produced CDs and huge, multi-channel surround-sound systems. High quality audio is no longer just the domain of the audiophile and the recording studio engineer, but live concert sound reinforcement as well. Advances in sound system technology treat concert-goers to shows that are bigger, louder, and offer greater clarity than ever before. The performer, however, is still dealing with a less-than-perfect solution for on-stage monitoring that was invented and, besides a few small exceptions, exploited to its practical limit in the Seventies. Traditional onstage monitor systems present myriad problems for the performer, as well as the audience, the FOH engineer, and the frazzled monitor engineer. Why should the artists be denied the opportunity to hear their performance in the best possible way?

“I am bowled over and fully converted to this exciting, well thought out product.”

*—Ian Anderson
of Jethro Tull*

Personal monitoring, also known as “in-ear,” is the logical solution. Formerly the province of high-end touring professionals, recent advances in wireless technology and universal earphone development have greatly reduced the initial investment costs for entrance into the world of personal monitors. As with most new technology, though, much misinformation and reluctance to experiment rise to the surface. In an effort to educate and enlighten anyone with an interest in improving the onstage listening experience, Shure presents this booklet, “The Shure Guide to Personal Monitors.” Divided into two sections and an appendix, the first section gives a short history of monitoring, the rise of personal monitors, and describes in detail the benefits of using them. Section two provides specific information on choosing the proper system to meet your needs, and the various ways personal monitor systems can be configured. The appendix includes additional information on wireless issues.

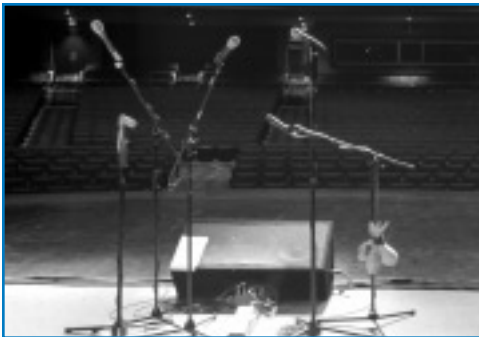
The field of personal monitoring is still growing and the technology continues to advance, so let us guide you through the possibilities and the advantages...

CHAPTER I

THE (BRIEF) HISTORY OF MONITORS

Although no one seems to know for sure, we can probably thank heavy metal bands from the late Sixties/early Seventies for necessitating the development of stage monitoring. Prior to the days of arena concerts and stacks of Marshall™ amplifiers, it wasn't that difficult to hear your voice coming through the main PA loudspeakers. Most concerts were held in smaller venues, with a few notable exceptions. When the Beatles played Shea Stadium, the only PA was for voice; guitars were only as loud as the guitar amplifiers. Of course, the crowd noise was so loud even the audience couldn't hear what was going on, let alone the band! As rock and roll shows continued to get bigger and louder, it became increasingly difficult for performers to hear what they were doing. The obvious solution was to turn some of the loudspeakers around so they faced the band. A further refinement came in the form of wedge-shaped speakers that could be placed on the floor, facing up at the band.

Besides being a convenient place for bass players to put their feet, wedge monitors finally gave singers the ability to hear themselves at a decent volume. With the size of stages increasing, it became difficult to hear everything, not just the vocals. The drums



Monitor Wedge

were on risers 15 feet in the air, and guitar amps were occasionally stowed away under the stage. These changes required the use of a monitor console – a separate mixer used for the sole purpose of creating multiple monitor mixes – to accommodate all the additional inputs as well as create separate mixes for each performer. Today, even the smallest music clubs offer at least two or three separate monitor mixes, and it is not uncommon for local bands to carry their own monitor rig capable of handling up to four mixes.



Professional Monitor Console

The problems created by traditional monitor systems are numerous; we'll take a detailed look at them in the next chapter. Suffice it to say, a better way to monitor needed to be found. Drummers have used headphones for years to monitor click tracks (metronomes) and loops. Theoretically, if all performers could wear headphones, the need for monitor wedges would be eliminated. Essentially, headphones were the first personal monitors – a closed system that doesn't affect or depend on the monitoring requirements of the other performers. Unfortunately, they tend to be cumbersome and not very attractive. Recent advances in the miniaturization of transducers have allowed performers to use "earphones," essentially headphones reduced to a size that fit comfortably in the ear. Professional musicians, including Peter Gabriel and the Grateful Dead, were among the first to employ this new technology. The other major contribution to the development of personal monitors is the growth of wireless microphone systems. Hardwired monitor systems are fine for drummers and keyboardists that stay relatively stationary, but other musicians need to be more mobile. Wireless personal monitor systems, essentially wireless microphone systems in reverse, allow the performer complete freedom of movement.

The first personal monitor systems were prohibitively expensive; only major touring acts could afford them. As with any new technology, as usage becomes more widespread, prices begin to drop. Current personal monitor systems have progressed to a point where they are within any performer's reach.

When was the last time you had a great experience with a wedge monitor system? You could hear everything, no feedback, plenty of volume (without being dangerous), and the monitor engineer instantly responded to your every request? Yeah – that’s what I thought. Anyone who has performed live has probably dealt with a poor monitor system, but even a great system has many limitations due to the laws of physics, and those laws bend for no one. The concept of “in-ear” monitoring rose from the desire to create an on stage listening experience that could overcome the limitations imposed by a traditional floor monitor system.

Let’s define a personal monitor system. There are many parallels between personal monitors and a traditional floor wedge setup. The purpose of any monitor system is to allow performers to hear themselves. The sounds to be monitored need to be converted to electronic signals for input to the monitor system. This is usually accomplished via microphones, although in the case of electronic instruments such as keyboards and electronic drums, the signals can be input directly to a mixing console. The various signals are then combined at a mixer, and output to either power amplifiers and loudspeakers or to the inputs of a personal monitor system. Any amount of signal processing, such as equalizers or dynamics processing (compressors, limiters, etc.), can be added in-between. A hardwired personal monitor system is similar (in signal flow terms) to a traditional wedge system, since the belt pack is basically a power amplifier, and the earphones are tiny loudspeakers.



PSM 600 (Hardwired)

A wireless personal monitor system, however, adds a few more components, specifically a transmitter and receiver. From the output of the mixer, the audio signal goes to a transmitter, which converts it to a radio frequency (RF) signal. A belt-pack receiver, worn by the performer, picks up the RF signal and converts it back to an audio signal. At this stage the audio is then amplified and output to the earphones. See Chapter 4 for a complete discussion of the various earphone types available.



PSM 400



PSM 700

The term “personal monitors” is derived from several factors, but basically revolves around the concept of taking a monitor mix and tailoring it to each performer’s specific needs, without affecting the performance or listening conditions of the others. The concept is broader than that of “in-ear” monitoring, which states where the monitors are positioned, but gives no further information on the experience.

So just what is the “experience” of personal monitors? The four most prominent benefits when using them are listed below:

- Superior sound quality
- Portability
- Mobility
- Personal Control

Superior Sound Quality

There are several factors that, when taken as a whole, result in the superior sound quality of personal monitor systems. These factors include adequate volume for the performers,

gain-before-feedback, hearing conservation, reduced vocal strain, and less interference with the audience mix.

Adequate Volume

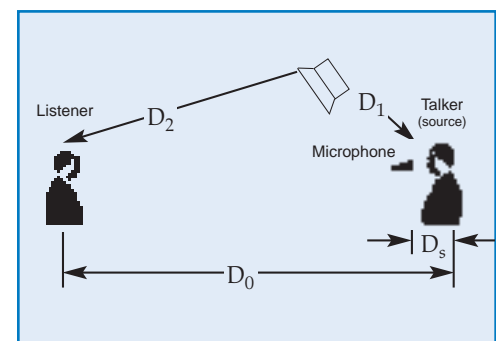
The most common request given to monitor engineers is “Can you turn me up?” (Sometimes not phrased quite so politely.) Unfortunately, it is not always quite that simple. There are many limiting factors to how loud a signal can be brought up when using traditional floor monitors: size of the power amplifiers, power handling of the speakers, and most importantly, potential acoustic gain (see Gain-Before-Feedback below). Another factor that makes hearing oneself difficult is the noise level onstage. Many times, vocalists rely solely on stage monitors, unlike guitarists, bassists, and keyboardists whose instruments are generally amplified to begin with. Drummers, of course, are acoustically loud without amplification. Volume wars are not uncommon as musicians struggle to hear themselves over the ever-increasing din. The clarity of the vocals is often obscured as other instruments are added to the monitor mix, which becomes increasingly necessary if fewer mixes are available. Keyboards, acoustic guitars, and other instruments that rely on the monitors often compete with the vocals for sonic space.

A personal monitor system, which isolates the user from crushing stage volumes and poor room acoustics, allows the musician to achieve a “studio-like” quality in the onstage listening experience. Shure E1 and E5 earphones, when used properly, provide between 10 and 20 dB of reduction in background noise level (see Chapter 4 for more information on the E1 and E5). The monitor mix can then be tailored to individual taste without fighting against otherwise uncontrollable factors.

Gain-Before-Feedback

When it comes to achieving higher monitoring levels with traditional stage wedges, you can always add more amplifiers and more loudspeakers, but you cannot defy the laws of physics. The concept of gain-before-feedback relates to how loud a microphone can be turned

up before feedback occurs. Closely related is PAG, or Potential Acoustic Gain. The PAG equation is a mathematical formula that allows you to predict how much gain is available in your sound system before reaching the feedback threshold, simply by plugging in known factors such as source-to-microphone distance and microphone-to-loudspeaker distance. Simply stated, the further away you get from the microphone, or the closer the microphone is to the loudspeaker, or the further away the loudspeaker is from the listener, then the less available gain-before-feedback.



Potential Acoustic Gain

$$PAG = 20 (\log D_1 - \log D_2 + \log D_0 - \log D_s) - 10 \log NOM - 6$$

Now picture a typical stage. The microphone is close to your mouth; that’s good. The microphone is close (relatively) to the monitor loudspeaker; that’s bad. The monitor loudspeaker is far (relatively) from your ears; that’s also bad. Feedback occurs whenever the sound entering a microphone is reproduced by a loudspeaker and “heard” by the same microphone again. To achieve a decent monitoring level, you need quite a bit of available gain. But given the above situation, you have two major factors working against you. Compounding the problem is the issue of NOM, or Number of Open Microphones. Every time you double the number of open microphones, the available gain-before-feedback drops by 3 dB. With four open microphones on stage instead of one, the available gain has dropped by 6 dB.

What can you do? The PAG equation assumes omnidirectional microphones, so using cardioid or even supercardioid pattern microphones will

“I was sold almost immediately on the Shure [PSM 600]. I no longer had to shout over a monitor. I felt more comfortable about my pitch, and I can listen at lower volumes, which means I never go home with ringing in my ears.”

—David Byrne

help; just don't point them at the speakers. Also, the equation assumes that the sound system has a perfectly flat frequency response. The most commonly employed tool for reducing feedback due to response problems is the graphic equalizer. Since some frequencies will feedback before others, an equalizer allows a skilled user to reduce the monitor system's output of those troublesome frequencies. This technique results in approximately 3-9 dB of additional gain, assuming the microphone position doesn't change. It is common practice for some monitor engineers to attempt to equalize the monitor system to the point where there is no feedback, even with a microphone pointed right into the speaker cone. Unfortunately, the fidelity of the monitor is often completely destroyed in an effort to eliminate feedback using equalizers. Even after equalization has "flattened" the response of the monitor system, PAG again becomes the limiting factor. At this point, you can't get any closer to the microphone, and moving the loudspeaker closer to your ears also makes it closer to the microphone, negating any useful effect on PAG.

Personal monitoring completely removes PAG and gain-before-feedback issues. The "loudspeakers" are now sealed inside your ear canal, isolated from the microphone. With the feedback loop broken, it is possible to achieve as much volume as necessary – which leads to the next topic...

Hearing Conservation

The main purpose of personal monitors is to hear yourself better. But it doesn't do any good if you can't hear at all. As mentioned earlier, volume wars on stage are a universal problem. Prolonged exposure to extremely high sound pressure levels can quickly cause hearing to deteriorate. Some performers have taken to wearing ear plugs to protect their hearing, but even the best ear plugs cause some alteration of frequency response. Personal monitors offer a level of hearing protection equal to that of ear plugs, but with the additional benefit of tiny loudspeakers in the plugs.

The monitoring level is now in the hands of

the performer. If it seems to be too loud, there is no excuse for not turning the monitors down to a comfortable level. The use of a limiter (standard on Shure PSM systems) is strongly recommended to prevent high level transients from causing permanent damage. In larger, complex monitor rigs, outboard compressors and limiters are often employed to offer a greater degree of control and protection.

NOTE: Using a Shure PSM system does not guarantee that you will not or can not suffer hearing damage. With the limiter switched off, these systems are capable of producing levels in excess of 130 dB SPL. Prolonged exposure to these kinds of levels can cause hearing damage. It is up to the individual user to be responsible for protecting his or her own hearing. If you have further questions or concerns about hearing conservation, contact a professional audiologist.

Reduced Vocal Strain

Closely related to the volume issue, the ability to hear more clearly reduces vocal strain for singers. In order to compensate for a monitor system that does not provide adequate vocal reinforcement, many singers will force themselves to sing with more power than is normal or healthy. Anyone who makes a living with their voice knows that once you lose it, you lose your livelihood. Every precaution should be taken to protect your "instrument," and personal monitors are a key ingredient in helping vocalists continue to sing for years to come. (See Adequate Volume, previously discussed.)

Interference with the Audience Mix

The benefits of personal monitors extend beyond those available to the performer. An unfortunate side-effect of wedge monitors is spill from the stage into the audience area. Although directional at high frequencies, speaker cabinets radiate low frequency information in a more or less omnidirectional manner. This situation aggravates the already complex task facing the FOH (front-of-house) engineer, who must fight against loud stage volumes when creating the audience mix. The excessive low frequencies coming off the backs

“It's the best system I have used to date. I was very impressed with the quality.”

*—Roger Daltrey
of The Who*

Stereo

A distinct advantage of using a personal monitor system is the ability to listen in stereo. While it may not be applicable to all situations, especially with a limited number of mixes available, a monitor mix created in stereo can more accurately recreate a realistic listening environment. We spend our entire lives listening in stereo; logically, a stereo monitor mix increases the perception of a natural sound stage.

Monitoring in stereo can also allow for lower overall listening levels. Imagine a group with two guitar players sharing the same mix. Both instruments are occupying the same frequency spectrum, and in order for each guitarist to hear, they are constantly requesting their own level be turned up. When monitoring in mono, the brain interprets sounds based only on amplitude and timbre. Therefore, when two sounds have roughly the same timbre, the only clue the brain has for perception is amplitude, or level. Stereo monitoring adds another dimension, localization. If the guitars are panned, even slightly, from center, each sound occupies its own “space.” The brain uses these localization cues as part of its perception of the sound. Research has shown that if the signals are spread across the stereo spectrum, the overall level of each signal can be lower, due to the brain’s ability to identify sounds based on their location.

of the monitors make the house mix sound “muddy” and can severely restrict the intelligibility of the vocals, especially in smaller venues. But eliminate the wedges, and the sound clears up considerably.

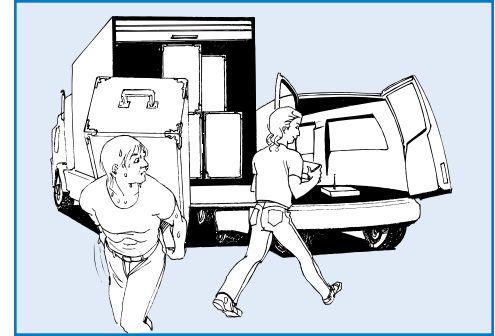
Portability

Portability is an important consideration for performing groups that travel, and for installations where the sound system or the band performance area is struck after every event. Consider the average monitor system that includes 3 or 4 monitor wedges at roughly 40 pounds each, and one or more power amplifiers at 50 pounds – this would be a relatively small monitor rig. A complete personal monitor system, on the other hand, fits in a briefcase.

Purely an aesthetic consideration, removing wedges and bulky speaker cables from the stage improves the overall appearance. This is of particular importance to corporate/wedding bands and church groups, where image is as important as sound. Personal monitors result in a very clean, professional-looking stage environment.

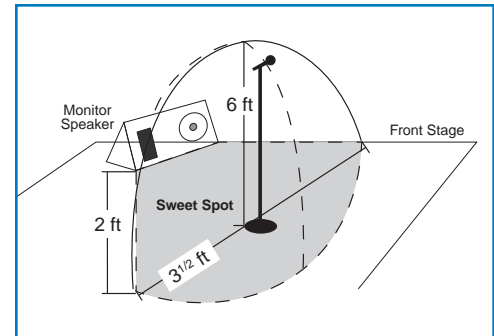
Mobility

Monitor wedges result in a “sweet spot” on stage; a place where everything sounds pretty good. If you move a foot to the left or right, suddenly things do not sound as good anymore. The relatively directional nature of loudspeakers, especially at high frequencies, is responsible for this effect. Using personal monitors, though, is like using headphones – the sound goes where you go. The consistent nature of personal monitors also translates from venue to venue. When using



Personal monitors won't break your back.

wedges, room acoustics play a large part in the overall quality of the sound. Since professional earphones form a seal against ambient noise, acoustics are removed from the equation. In theory, given the same band with the same members, the monitor settings could remain virtually unchanged, and the mix will sound the same every night.



Sweet spot created by a monitor wedge

Personal Control

Perhaps the most practical benefit to personal monitors is the ability to have direct control over what you are hearing. While still relying on the sound engineer to make fine adjustments, personal monitor systems give the performer some ability to make coarse adjustments, such as overall volume, pan, or the ability to choose different mixes. If everything in the mix needs to be louder, instead of giving a series of complex hand gestures to the monitor engineer, the performer can raise the overall volume directly from the belt-pack.

Use of the MixMode™ feature on Shure PSM systems expands the level of personal control. MixMode™ utilizes a “dual-mono” scheme, where the belt-pack combines the left and right audio channels and sends the combined

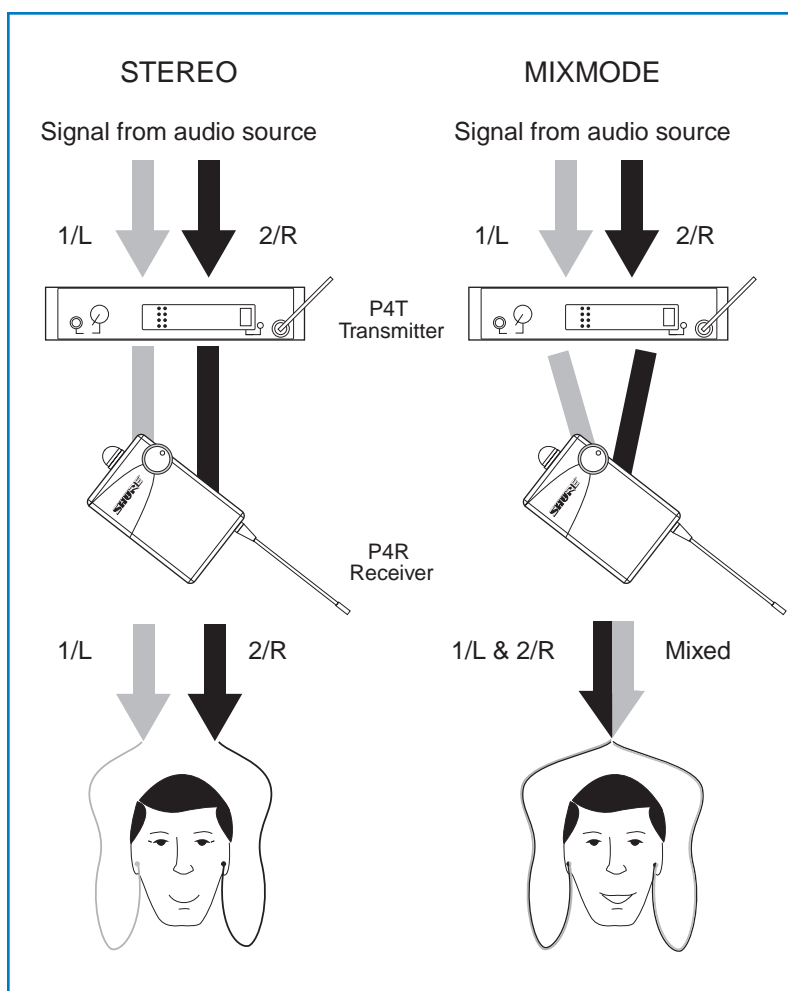
signal to both sides of the earphones. The inputs to the system should now be treated as “Mix 1” and “Mix 2” instead of left and right. The balance control on the receiver now acts as a mix control, allowing the performer to choose between two mixes, or to listen to a combination of both mixes with control over the level of each. Panning to the left gradually increases the level of “Mix 1” in both ears, while reducing the level of “Mix 2,” and vice versa. Chapter 5 includes some practical applications for MixMode™.

Putting a small, outboard mixer, such as the Shure P4M (see Chapter 6), near the performer increases the amount of control. By giving control of the monitor mix to the performer, the sound engineer can spend more time concentrating on making the band sound good for the audience instead of worrying about making the band happy.

The cost of transitioning to personal monitors has recently dropped dramatically. A basic system costs as much, if not less than, a typical monitor wedge, power amplifier, and graphic equalizer combination. Expanding a system is also more cost effective. When providing additional wedges for reproducing the same mix, a limited number can be added before the load on the amplifier is too great, and another amp is required. With a wireless personal monitor system, however, the number of receivers monitoring that same mix is unlimited. Additional receivers do not “load” the transmitter, so feel free to add as many receivers as necessary without adding more transmitters. For bands that haul their own PA, transportation costs may be reduced as well. Less gear means a smaller truck, and possibly one less roadie.

“Shure personal stereo monitors make conventional stage monitors a thing of the past.”

—Sebastian Bach



Graphic representation of MixMode

Given the personal nature of “in-ear” monitoring, choosing the right system is an important step. Several choices are available, and present as well as future needs should be taken into account before making an investment. Use the following questions to help determine which system is right for you:

Do I Need a Wireless or a Hardwired System?

Personal monitor systems come in two basic varieties – wireless or hardwired. A hardwired system requires the performer be tethered to a cable, which is not necessarily a negative. Drummers and keyboard players who stay relatively stationary, or even back-up singers, can take advantage of the lower cost and greater simplicity of a hardwired personal monitor system. Simply connect the monitor sends to the inputs of the PSM hardware and dial up a mix. Hardwired systems also work worldwide without the hassle of finding clear frequencies or dealing with local wireless codes. Lastly, if several performers require the same mix, hardwired PSM systems can be daisy-chained together without significant signal loss. The input impedance of PSM hardwired systems is sufficiently high to allow multiple systems to be connected to a single output with Y-cables. At least eight systems can be connected to one output without any negative side effects. Another way to connect more than one hardwired system to the same output is to use a distribution amplifier, such as the Shure FP16A. A distribution amplifier takes a single input and splits it to multiple outputs, each with its own individual level control.



Distribution Amplifier (FP16A)

Wireless equipment, by nature, requires special considerations and attention to detail (see Appendix Wireless). But the advantages many times outweigh the increased cost and complexity. One of the main benefits of personal monitors is a consistent

mix no matter where you stand; going wireless allows you to exploit this advantage to its fullest extent. (On their 1998 Bridges to Babylon Tour, the Rolling Stones used PSM 600 to monitor themselves from a remote stage in the middle of the seating area!) Additionally, when several performers require the same mix, hooking them up is even easier. As many wireless receivers as necessary can monitor the same mix with no adverse effects. And of course, no cables to trip on!

How Do I Decide Between Shure PSM 400, 600, or 700?

This question can also be addressed in two parts – wired and wireless.

When choosing a hardwired personal monitor system, there are two choices: P4HW (PSM 400) or P6HW (PSM 600). The two systems share many of the same features, including mono, stereo, or MixMode™ operation. Both systems also have a switchable limiter, high frequency EQ boost, and a 15 dB input pad. The P4HW body pack is made of rugged Armo-Dur plastic, but if a metal body pack is required, choose the P6HW. The user interfaces are different as well. The controls for the P6HW (except for volume and pan) are hidden behind the battery compartment door and are adjusted via dip switches. The P4HW features a musician-friendly LCD display with Scroll and Select buttons to make adjustments. The inputs for both systems are on XLR connectors, but the P6HW uses a detachable cable that connects to the body pack with a metal LEMO connector, while the P4HW's cable is permanently attached. The P4HW includes XLR male to 1/4" male phone plug adapters to connect to mixers which only have 1/4" auxiliary outputs.

Note: There is no hardwired PSM 700 system.

For wireless applications, the selection process gets a little more complicated. The first consideration is the number of mixes required by your application, and whether they need to be stereo or mono. There is a direct correlation between the number of required mixes and the number of transmitters that can be operated

simultaneously in the same location. The different ways these mixes can be configured are covered in Chapter 5, so here are just the hard numbers:

- PSM 400 – 8 transmitters (8 stereo, 16 mono)
- PSM 600 – 10 transmitters (10 stereo, 20 mono)
- PSM 700 – 16 transmitters (16 stereo, 32 mono)

Secondly, consider the travel requirements, if any, of the users. Shure PSM systems transmit on unused television channels. Since these unoccupied channels will be different in every city, it is imperative that you choose appropriate frequencies. For a band that plays only in one metropolitan area, or for a permanent installation, once a good frequency is chosen, there should be no need to change it. However, for touring acts, the ability to change operating frequencies is almost essential. Luckily, all Shure PSM systems offer some degree of frequency agility.

The 10 available frequencies for PSM 600 are divided into five pairs of frequencies. Therefore, if one frequency is not clear, flip to the other and see if it is any better. The two frequencies are on adjacent TV channels (i.e. channel 40 and 41), so you can choose your frequency based on local television broadcast. PSM 600 works well for most fixed installations, as well as acts that only do a limited amount of touring. For extensive touring applications, especially when using more than five transmitters, a system with more frequency agility may be desired. Using the PSM 600 to its fullest extent (10 frequencies) requires *both* frequencies in each P6T transmitter be used. With all ten frequencies in operation, any frequency agility is lost. The Shure PSM 700 system ups the ante by including two groups of 16 compatible frequencies, or 32 total frequencies to choose from. With so many frequencies available, no matter where the tour ends up there should be something that will work. Even if you are using the maximum number of frequencies (16), you still have another group of 16 to switch to if necessary. PSM 700 also features a Frequency Locator Mode, which allows

the user to search for clear frequencies before the performance.

Of course, budget is always a consideration when purchasing new equipment. For those users who require frequency agility, but can not afford PSM 700, the PSM 400 is a viable alternative. A PSM 400 wireless system gives the user 16 frequencies to choose from, of which any eight are compatible. Changing frequencies on the P4T is accomplished via a recessed push-button switch that can be actuated with a 1/4" phone plug (which there should be plenty of hanging around the stage) instead of requiring a small screwdriver.

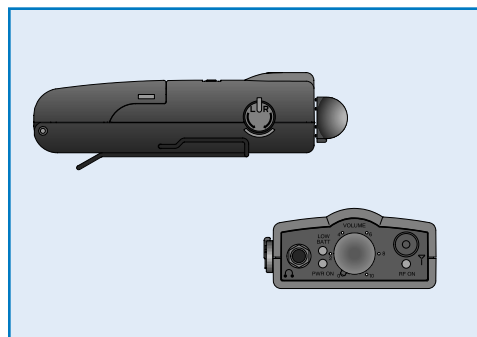
**Note: The available frequencies and number of compatible frequencies are country dependent.*

Why Would I Choose PSM 600 Over PSM 400?

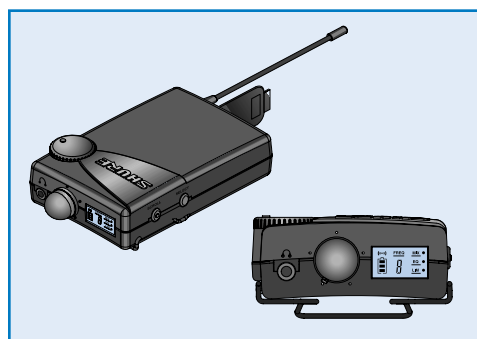
A valid question, given the less than obvious answer. On the surface, PSM 400 appears a better value, given the greater frequency agility at a lower cost. PSM 600, though, offers several professional features not present on PSM 400. (After all, it was good enough for the Rolling Stones!) Similar to the hardwired systems, the P6R receivers are metal instead of plastic, and the P6R employs hidden dip switches instead of an LCD display for the user interface. The P6T transmitters also have XLR-1/4" combination connectors on the inputs, internal power supplies, and removable antennas. The ability to remove antennas allows for antenna

“I could sing
night after
night and
not hurt
my voice.”

—LeAnn Rimes



PSM 600 Receiver (P6R)

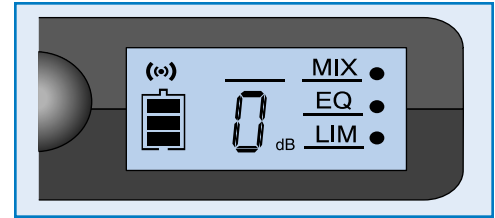


PSM 400 Receiver (P4R)

CHAPTER 3

CHOOSING THE RIGHT SYSTEM

combining (highly recommended for multiple transmitter setups) and remote antenna placement. Other PSM 600 advantages include an input level control, input sensitivity switch (+4 dBu or -10 dBV), and more compatible frequencies than PSM 400 (10 versus 8).



PSM 400 LCD Screen

Personal Stereo Monitor System Selection Guide (U.S.)

	PSM 400	PSM 600	PSM 700
Wireless Version	UHF 16 Selectable Frequencies across 4 TV Channels <i>Frequency Synthesized</i>	UHF 2 Selectable Frequencies across 2 TV Channels <i>Crystal</i>	UHF 32 Selectable Frequencies Across 4 TV Channels <i>Frequency Synthesized</i>
Frequency Range	722-746 MHz (TV 56-59)	626-662 MHz (TV 40-45)	722-746 MHz (TV 56-59)
Max Number of Transmitters in One Location¹	8	10	16
Transmitter Output Power	50 mW	100 mW	100 mW
Transmitter Input Level Control	No	Yes	Yes
Transmitter Input Connectors	1/4" balanced phone jacks	Combination XLR female – 1/4" phone jacks	Combination XLR female – 1/4" phone jacks
Transmitter Headphone Monitor Jacks	3.5 mm mini-phone jack	1/4" and 3.5 mm phone jacks	1/4" and 3.5 mm phone jacks
Transmitter Input Pad	No	Yes (-10 dBV or +4 dBu)	Yes (-10 dBV or +4 dBu)
Loop Through Outputs	Yes	Yes	Yes
Receiver Chassis	<i>High-Impact Plastic</i>	Metal	Metal
Receiver Battery Type/Life²	9-volt alkaline 8 hours	9-volt alkaline 6 hours	9-volt alkaline 6 hours
Transmitter Antennas	Fixed 1/4-wavelength	Removable 1/4-wavelength	Removable 1/4-wavelength
Rack Mount Hardware	Included	Included	Included
Antenna Combiner	N/A (<i>Antenna not removable</i>)	PA760	PA770
Antenna Cable	N/A	PA725 (10 ft.)	PA725 (10 ft.)
Remote Directional Antenna	N/A	PA705	PA705
Supplied Earphones³	E1	E1	E1/E5
Transmitter Power Supply	PS40	Internal	Internal
Hardwired Version	Yes	Yes	No
Hardwired Chassis Type	<i>High-Impact Plastic</i>	Metal	N/A
Hardwired Battery Type/Life²	9-volt alkaline 8 hours	9-volt alkaline 10 hours	N/A

¹Location Dependent

²Battery life is volume level dependent

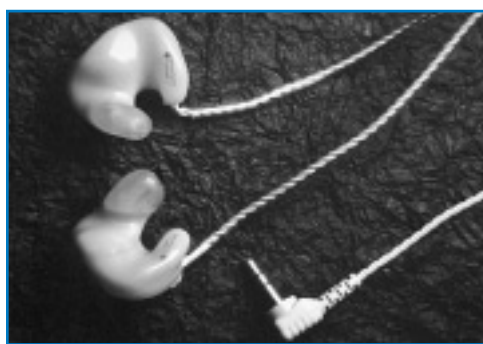
³E5 Earphones compatible with PSM systems

“This is really good sounding stuff, works great, held up nice... I like it.”

—J.D. DuCrest,
Monitor Engineer
for LeAnn Rimes

Earphone Options

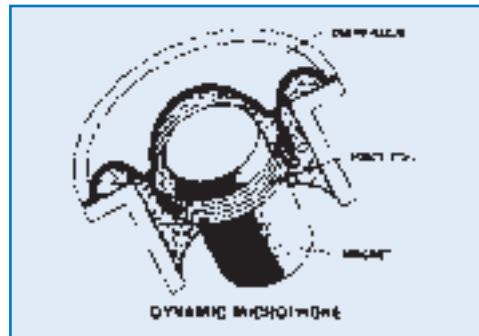
The key to successful personal monitoring lies in the quality of the earphone. All the premium components in the monitoring signal path will be rendered ineffective by a low quality earphone. A good earphone must combine full range audio fidelity with good isolation, comfort, and inconspicuous appearance. The types of earphones available range from inexpensive Walkman®-type “ear-buds” to custom molded, highly personalized designs. Each type has its advantages and disadvantages. While relatively affordable, “ear-buds” have the poorest isolation, and are not really designed to withstand the rigors of a working musician’s environment (not to mention they don’t stay in your ears when you’re headbanging!). On the other end of the spectrum, custom molded earphones offer exceptional sound quality and isolation, yet are extremely expensive, and are impossible to try before buying. The procedure for getting custom molds involves a visit to an audiologist. The audiologist makes an impression of your ear canals by placing a “dam” inside your ear to protect your ear drum, and fills them with a silicone based material that conforms exactly to the dimensions of your ears. The impressions are then used to create the custom molded earphones. Another visit to the audiologist is required for a final fitting. Manufacturers of custom molded earphones include Ultimate Ears and Sensaphonics.



Custom molded earphone
(SP 2000 courtesy Sensaphonics)

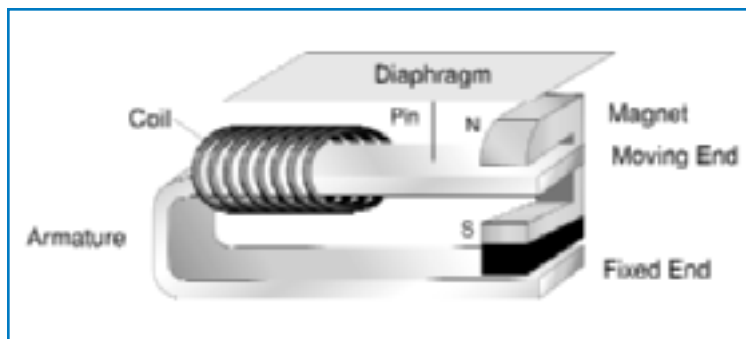
The internal workings of earphones vary as well. There are two basic types of transducer used in earphone design – dynamic and balanced armature. The dynamic types work

on the same principle as dynamic microphones or most common loudspeakers. A thin diaphragm is attached to a coil of wire suspended in a magnetic field. Diaphragm materials include Mylar (in the case of dynamic microphones) or paper (for loudspeakers). As current is applied to the coil, which is suspended in a permanent magnetic field, it vibrates in sympathy with the variations in voltage. The coil then forces the diaphragm to vibrate, which disturbs the surrounding air molecules, causing the variations in air pressure we interpret as sound. Dynamic transducers are used mainly in the “ear-bud” types, although they are also found in some custom molded earphones.



Dynamic Transducer

Originally implemented in the hearing aid industry, the balanced armature transducer combines smaller size with higher sensitivity. A horseshoe shaped metal “arm” has a coil wrapped around one end and the other suspended between the north and south poles of a magnet. When alternating current is applied to the coil, the opposite arm (the one suspended in the magnetic field) is drawn towards either pole of the magnet. The vibrations are then transferred to the diaphragm, usually a thin layer of foil. Balanced armature transducers are similar to the elements used in controlled magnetic microphones.



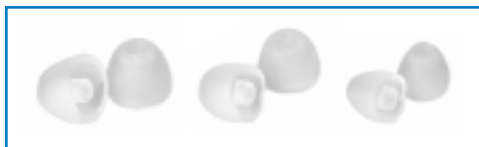
Balanced Armature Transducer

E1

In 1997, Shure introduced the E1, an isolating “universal” earphone designed for in-ear monitoring. A low-mass, high-energy driver that uses balanced armature technology, the E1 was created to combine the superior isolation and fidelity of custom molded designs with the out-of-the-box readiness of “ear-buds.” The “universal” nature of the E1 is attributed to the interchangeable sleeves that are used with the standard E1. In other words, the user has his or her choice of earphone sleeves that fit any E1. This design allows the user to audition the various sleeves to see which works best for them, as well as being able to demo the earphones before a purchase is made. The different ear piece sleeves include foam, flex sleeves, rubber flange tips, and custom molded. Also, if there is ever a problem with the E1, another set can be substituted with no negative repercussions. A custom molded earphone does not allow for this kind of versatility; if one needs repair, the only alternative is to have a back-up (at roughly \$500 - \$750 a pair!) to use in the interim.



E1 Earphones with PA750



E1 Earphone Sleeves (PA756)

E5

The second generation of Shure “universal” earphones is the E5. Designed for high-end applications, the E5 improves upon the sound quality of the E1 by implementing a dual-transducer (or dual-driver) design. Another

example of a loudspeaker with dual-transducer design is one with a horn (or tweeter) for high-frequency reproduction and a woofer for low-frequency sounds. The frequency spectrum is divided in two by a crossover network. Each driver only has to reproduce the frequency range for which it has been optimized. The E5 works on a similar principle – each earphone contains a “tweeter” and a “woofer” optimized for high and low frequency performance, respectively. Additionally, a passive crossover is built into the cable to split the frequencies (at roughly 3 kHz). The end result is usually much more low end, as well as extended high frequency response. The E5 may be of particular interest to bassists and drummers, whose instruments produce a good deal of low frequency content. The same sleeve options are available for both the E1 and the E5. Note that the custom sleeves for each are different, though. You cannot use sleeves designed for the E1 with the E5, and vice versa.



E5 Earphones

Earphone Sleeves

Foam – PA750

The most common of the sleeve options: the foam sleeves resemble regular foam ear plugs, but with a small hole in the center of the foam lined with a tube of plastic. The shaft of the earphone is inserted through this hole, shiny side facing out. To properly seat the E1 (or E5) in your ear, compress the foam as small as possible, and insert the earphone into your ear canal. Hold it in place until the foam expands completely, forming a tight seal. Drape the cable over the top of your ear. Repeat this procedure for your other ear, and then use the plastic adjustment tube on the cable to reduce

“It just sounds so much better. Live is very loud on stage, and Ed (Kowalczyk) was always fighting to hear his monitors. It was a major battle. When I got the PSM 600 to try, it was amazing, hearing it through my own custom molds.”

—Brendan McCabe,
Monitor Engineer
for Live

excess slack. (The earphones are marked with a blue dot for the left ear, and red for the right.) The foams offer excellent isolation and good low frequency performance. On the downside, they eventually get dirty and worn, and need to be replaced. Proper insertion of the foams also takes longer – relative to the other options – since you need to hold the earphone in place while the foam expands.

Flex Sleeves – PA756 (S, M, or L)

For quick insertion and removal of the earphones, the flex sleeves may be a good choice. Made of soft, flexible plastic, the flex sleeves resemble a mushroom cap and are available in three sizes (small, medium, and large). Insertion is very easy, just place the sleeve in your ear canal and drape the cable as detailed above. While the seal is usually not as tight as with the foams, the flex sleeves are washable and reusable.

Triple-flange Sleeves – PA755

Similar to the flex sleeves, the triple-flange sleeves have three rings (or flanges) around a central rubber tube. They are sometimes referred to as “Christmas trees” based on their shape. The pros and cons are similar to that of the flex sleeves, but they have a different comfort factor that some users may find more to their liking.

Custom Sleeves

The fourth, and most expensive, option is custom sleeves. The custom sleeves combine the relative ease of insertion and permanency of the flex sleeves with the superior (depending on the preference of the user) isolation of the

foams. The process for obtaining custom sleeves for Shure earphones is very similar to that of getting custom molded earphones; a visit to an audiologist is required to get impressions made of your ear canals. Custom sleeves also give the user many of the same benefits as custom molded earphones, but usually at a lower cost, and with the added benefit of being able to interchange earphones with the sleeves if they get lost, stolen, or are in need of repair.

IMPORTANT NOTE: There are several brands of custom molded earplugs with internal filters that have relatively flat frequency response and different levels of attenuation. Although it is physically possible to make Shure earphones fit into the plugs with the filter removed, we strongly recommend against it. The location of the E1 or E5 in the ear canal is crucial to obtaining proper frequency response, and most earplugs will prevent the earphones from getting in the proper position. Once again, custom molded earplugs are NOT an acceptable alternative to custom sleeves.

Supplementary Equipment

In-ear monitoring is a different auditory experience from traditional stage monitoring. Since your ears are isolated from any ambient sound, the perception of the performance environment changes. There are several other types of audio products that can be added to a personal monitor system to enhance the experience, or try to simulate a more “live” feel.

Drum Throne Shakers

Something performers may miss when making the transition to personal monitors is the vibrations created by amplified low frequency sounds. Drummers and bass players are particularly sensitive to this effect. While moving to an earphone like the E5 will definitely result in more perceived bass, an earphone cannot replicate the sensation of physical motion. Drum shakers exist not to provide any audible sound reinforcement, but to recreate the vibrations normally produced

“Let me just rave to you a bit about Shure’s PSM 600 system. The on-board limiter is wonderful, the E1 Earphones work great out of the box, and the audio quality is excellent. Having Marty’s drummer on your system allows me to deliver more consistent, high quality sound to a critically demanding player.

This is a fine product.”

–Les Banks,
Monitor Engineer
for Marty Stewart



Triple-Flange Sleeves (PA755)

by subwoofers or other low frequency transducers. Commonly found in car audio and cinema applications, these devices mechanically vibrate in sympathy with the musical program material, simulating the air disturbances caused by a loud subwoofer. They can be attached to drum thrones or mounted under stage risers.

Ambient Microphones

Ambient microphones are occasionally employed to restore some of the “live” feel that may be lost when using personal monitors. They can be used in several ways. For performers wishing to replicate the sound of the band on stage, a couple of strategically placed condenser microphones can be fed into the monitor mix. Ambient microphones on stage can also be used for performers to communicate with one another, without being heard by the audience. An extreme example (for those whose budget is not a concern) is providing each performer with a wireless lavalier microphone, and feeding the combined signals from these microphones into all the monitor mixes, but not the main PA.

Shotgun microphones aimed away from the stage also provide good audience pick-up, but once again, a good condenser could suffice if shotguns are not available.

Effects Processing

Reverberant environments can be artificially created with effects processors. Even an inexpensive reverb can add depth to the mix, which can increase the comfort level for the performer. Many singers feel they sound better with effects on their voices, and in-ear monitors allow you to add effects without disturbing the house mix or other performers. A note of caution: any digital processor adds a certain amount of latency in the signal path. For example, routing a signal through a digital effects processor requires time to convert the signal from analog to digital, process the

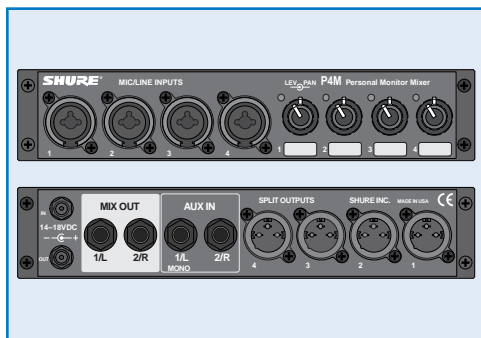
signal, and convert it back to analog. Latency is the time it takes to accomplish this processing. The degree of latency is generally not more than a few milli-seconds, which will not cause the processed signal to be perceived as an audible delay. The concern for users of in-ear monitors, though, lies primarily with horn players, and occasionally vocalists. When a horn player sounds a note, the vibrations are carried directly to the ear canal via bone conduction. If the microphone signal is subject to digital processing, too much latency can cause comb filtering. The user generally perceives this as a hollow, unnatural sound. Care should be taken to avoid introducing unnecessary processing if comb filtering occurs. Adjusting the delay time in the processor (assuming digital delay is one of the available effects) could also compensate for latency.

Outboard compressors and limiters can also be used to process the audio. Although all Shure PSM systems have a built-in limiter, external limiters will provide additional protection from loud transients. Compression can be used to control the levels of signals with wide dynamic range, such as vocals and acoustic guitar, to keep them from disappearing in the mix.

The Occluded Ear

One final note for users who are new to earphones. When your ear canal is acoustically sealed (“occluded”), the auditory experience is different from what you may be traditionally accustomed to. For those performers who have spent many years using traditional floor monitors, an adjustment period may be necessary. A common side effect for vocalists is under-singing. The sudden shock of hearing yourself without straining causes some vocalists to sing softer than they normally would, making it difficult for the FOH engineer to get the vocals loud enough in the house mix. Remember, the FOH engineer is still fighting the laws of PAG, so you need to project!

Hooking up PSM systems and making them work is a relatively simple process, but the ways in which they can be configured are almost limitless. In this section, we'll take a look at several typical system set-up scenarios. Personal monitor systems are equally useful for performance and rehearsal, and their benefits extend from small nightclub settings to large arena tours to houses of worship.



P4M

Using PSM for Rehearsals

If you already own a mixer, implementing a PSM system for rehearsals is a simple process. There are a number of ways you can get signal into the system, depending on how many mixes are necessary. To create a simple stereo mix, simply connect the main outputs of the mixer directly to the PSM inputs. Auxiliary sends can also be used if separate mixes are desired. For bands that carry their own PA system (or at least their own mixer), this method allows them to create a monitor mix during rehearsal, and duplicate it during a performance. No adjustment needs to be made for the acoustic properties of the performance environment.

Since many performers don't own their own mixer, the P4M can be used to create mixes for rehearsals. See Chapter 6 for examples of implementing P4Ms in rehearsals. When it comes to performance time, your monitor mix can be dialed-in exactly as it was in rehearsal, without affecting the house mix.

Using PSM for Performance

Club/Corporate/Wedding Bands – No Monitor Mixer

The majority of performing groups do not have the benefit of a dedicated monitor mixer. In this situation, monitor mixes are created using the auxiliary sends of the main mixing console. The number of available mixes is limited by the capabilities of the mixer (see the AuxPander in Chapter 6). At its most basic level, a PSM system can provide at least one stereo or two mono mixes. Therefore, any mixer should be capable of providing at least two dedicated, prefader auxiliary sends. Prefader sends are unaffected by changes made to the main fader mix. Postfader sends change level based on the positions of the channel faders. They are usually used for effects. Although postfader sends can be used for monitors, it can be somewhat distracting for the performers to hear fader moves.

Most users that only have two auxiliary sends available will probably choose MixMode™ operation, since this allows for the most flexibility. Hookup is easy – just connect Aux Send 1 of your console to the left input of the PSM transmitter (or hardwired system) and Aux Send 2 to the right input. (Use Aux 3 and 4 if those are the prefader sends – all consoles are different!) Then, depending on who is listening to what, create the mixes by turning up the auxiliary sends on the desired channels. A few common two-mix setups are listed below:

Two Monitor Mixes (MixMode™)

Option 1

Aux 1 (PSM Left)	Aux 2 (PSM Right)
Vocal Mix	Band mix (Guitars, drums, etc.)

Option 2

Aux 1 (PSM Left)	Aux 2 (PSM Right)
"Front" mix (Vocals, guitars, horns, etc.)	"Backline" mix (Drums, bass)

Option 3

Aux 1 (PSM Left)	Aux 2 (PSM Right)
"Ego" mix (Lead instrument or vocal)	Everything else

*“I don’t
ever want
to play
without it.”*

*–Michael Baker,
Drummer for
Whitney Houston*

CHAPTER 5

SETTING UP A PERSONAL MONITOR SYSTEM

Each performer can choose which mix they want to listen to by adjusting the balance control on the receiver. Be sure the receivers are set for MixMode™ operation, or each mix will only be heard on the left or right side, but not in both ears. Also remember that any number of receivers can monitor the same transmitter.

Some performers may prefer to listen to the house mix, so they can monitor exactly what the audience is hearing. Rather than try to duplicate the house mix with auxiliary sends, the main outputs of the console can be connected directly to the PSM transmitter. The Loop Out jacks allow the signal to pass on to the main PA, unaffected by the monitor system. Keep in mind that this will not always

produce the desired results. Rarely will what sounds good in the ear canal sound equally as good through a PA system in a less-than-perfect acoustic environment. Many times, a vocal that seems to blend just right for an in-ear mix will get completely lost through the PA, especially in a small room when live instruments are used. This technique may be appropriate for electronic bands, where the majority of instruments are input directly to the mixer. The only sound in the room is that created by the sound system.

The more auxiliary sends a console has, the more monitor mixes you can create. See the tables below for more examples.

Recommended systems: PSM 400 or PSM 600

Three Monitor Mixes (MixMode™)

Option 1			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	PSM 2 Right
Vocal mix	Band mix	Dedicated drum mix	Unused

Option 2			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	PSM 2 Right
Lead vocal	Everything else	Dedicated drum mix	Unused

Option 3			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	PSM 2 Right
"Front" mix	"Backline" mix	"Ego" mix (bandleader gets whatever he/she wants!)	Unused

Four Monitor Mixes (MixMode™ – using only 3 Aux Sends and PSM Loop Jacks)

Option 1			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	PSM 2 Right
Vocal mix	Band mix	Horn mix	Band mix (looped from PSM Right Loop Out Jack)

Four Monitor Mixes (MixMode™)

Option 1			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	Aux 4 Out (PSM 2 Right)
Lead Vocalist's mix	Guitarist's mix	Bassist's mix	Drummer's mix

Option 2			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	Aux 4 Out (PSM 2 Right)
Vocal mix	Band mix	Horn mix	Vocal/Band mix

Option 3			
Aux 1 Out (PSM 1 Left)	Aux 2 Out (PSM 1 Right)	Aux 3 Out (PSM 2 Left)	Aux 4 Out (PSM 2 Right)
"Ego" mix (lead vocal/instrument only)	"Ego" mix (everything else)	Band mix	Dedicated Drum mix

Club Level Bands – with monitor console

At this point, we've reached the limit of monitoring capabilities for the typical small format FOH console. Bands that have graduated to the next level of performance (larger, more prestigious clubs and theaters or small tours) may find themselves in a position to take advantage of a dedicated monitor console. Most monitor boards are capable of providing at least eight mono (or four stereo) mixes. It now becomes practical for each band member to have his or her own dedicated mix. System hookup is again very simple - the various mix outputs from the monitor console are connected directly to the PSM system. Stereo monitoring is a much more viable option due to the large number of mixes available, as well as the presence of a skilled monitor engineer (hopefully!) to tweak the mixes to the point of perfection.

Some performers even carry their own monitor console. Due to the consistent nature of personal monitors, a band with the same instrumentation and performers for every show can leave the monitor mix dialed-in on their console. Since venue acoustics can be completely disregarded, a few minor adjustments are all that is typically necessary during sound check.

A personal monitor mixer can also be used to augment the monitor console, if the performer desires some personal control over what they hear. In the past, drummers or keyboard players would use a small mixer and Y-cables to submix their instruments for in-ear monitors. The Shure P4M Personal Monitor Mixer can be used in the same capacity, but without the need for Y-cables. See Chapter 6 for more on the P4M.

A note on monitor mixing: Performers now have an unprecedented level of personal control over what they are hearing. The temptation to make yourself the loudest thing in the mix is great, but this may not be the best for the situation. Proper blending with the other members of your ensemble will be next to impossible if the mix is skewed too far from reality. Consider big bands that normally play

acoustically, or a vocal choir. These types of ensembles create their blend by listening to each other, not just themselves. If the lead trumpet player uses a personal monitor system, and cranks the trumpet up three times louder than everything else, there is no accurate feedback for the musician on whether they are playing too loud or too soft. Remember, great bands “mix” themselves – don't rely entirely on the sound tech to get it right.

Recommended systems: PSM 600 or PSM 700

Professional Touring System

When budget is no longer a consideration, personal monitoring can be exploited to its fullest capabilities. Many systems used by professional artists on large scale tours often employ greater than 16 stereo mixes.

A completely separate, totally personalized mix is provided for every performer on stage. Large frame monitor consoles are a requirement. For example, to provide 16 stereo mixes requires a monitor console with 32 outputs. Effects processing is generally employed to a much larger extent than with a smaller system.

When operating a large number of wireless personal monitor systems, RF related issues become much more important (Appendix). Frequency coordination must be done carefully to avoid interaction between systems as well as outside interference.

Depending on the extent of the touring, a frequency agile system is desirable, if not required. Proper antenna combining, to reduce the number of transmitter antennas in close proximity, is a necessity. Directional antennas may also be used to increase range and reduce the chances of drop-outs due to multipath interference.

Recommended systems: PSM 600 or PSM 700

“I love it. It sounds great, much better than our old systems. When you go from a huge outdoor gig to a small theatre, your sound on stage remains the same.”

–Kris Umezawa,
Monitor Engineer
for David Byrne



N' Sync

Personal Monitors for Houses of Worship and Sound Contractors

The advantages of using personal monitors extend beyond those of just the performers. We have seen how they benefit the performer, and up to this point we have been discussing personal monitors from a strictly music industry oriented point of view. This section will discuss how personal monitors can be a useful tool for the sound contractor, specifically as they apply to modern houses of worship.



Musical performances are rapidly becoming a more prominent part of the worship service. Praise teams and contemporary music groups, while bringing new levels of excitement to traditional church services, also bring with them the problems

of your average rock band. Most prominent among these problems are volume wars. Drums naturally tend to be the loudest thing on stage. The guitarist, in order to hear himself better, turns his amplifier up louder. The singers then need more monitor level to compete with the rest of the band. And then the cycle begins again. In any live sound situation, church or otherwise, loud stage volumes can distract from the overall sound in the audience. Try an easy experiment at the next sound check. When the band is satisfied with the monitor mix (such as it is...), turn off the audience PA and just listen to the sound coming off the stage. It's probably loud enough that you don't even need to turn on the main system! To compound matters, the "backwash" off the floor monitors consists primarily of low frequency information that muddies-up the audience mix. While this situation creates headaches for most sound engineers, it is even worse in the church environment. The majority of Sunday morning service attendees are not looking for an extremely loud rock and roll concert, but in some cases the congregation mix gets this loud just so it can be heard over the stage monitors. If you turn

off the main system and it's still too loud, what can you do? If you turn down the floor monitors, the band complains – not to mention how terrible it will sound.

With the band using personal monitors, these problems evaporate. Traditional floor monitors can be completely eliminated. For part two of our experiment, turn off the stage monitors while the band is playing. Notice how much clearer the audience mix becomes? This is how it would sound if the band were using personal monitors. Also, personal monitors are not just for vocalists. Drummers with in-ear monitors tend to play quieter. When the loudest instrument on stage gets quieter, everything else can follow suit. Some churches take this a step further by using electronic drums which create little, if any, acoustic noise. Bass, keyboard, and electric guitar can also be taken directly into the mixer if the players are using personal monitors, eliminating the need for on-stage amplifiers. The end result is a cleaner, more controlled congregation mix, and musicians can have very loud monitors without affecting the congregation.

Secondly, consider the feedback issue. Feedback occurs when the sound created at the microphone comes out of a loudspeaker, and reenters the microphone. The closer the loudspeaker is to the microphone, the greater the chance for feedback. By eliminating the floor monitor, you also eliminate the worst possible feedback loop. With the "loudspeakers" sealed inside the ear canal, there is no chance for the signal to reenter the microphone. No equalizer or feedback reducer will ever be as effective as personal monitors will at eliminating feedback on the stage.

Many other uses are possible for personal monitors. Choir directors could use them for cues, or to hear the pastor more clearly. Organists located at the rear of the sanctuary could use them to better hear the choir located up front, or also to receive cues. The advantages extend well beyond the benefits to the performer, and increase the overall quality of the service and the worship experience.

Recommended systems: PSM 400 or PSM 600

Personal Monitor Mixers

Personal monitoring gives the performer an unprecedented level of control. But for the performer who desires more than simple volume and pan (or MixMode™) operation, an additional mixer can be implemented. Personal monitor mixers are especially useful for bands who have a limited number of available monitor mixes, or who do not have a monitor engineer, or anyone at all to run sound. In a perfect world, all performers would be happy listening to the exact same mix; in reality, everyone may want to hear something different. A small mixer located near the performers allows them to customize their mix to hear exactly what they desire. Theoretically, any mixer can double as a personal monitor mixer, but most lack one key feature; the input signals need to find their way to the main (FOH) mixer somehow. Large sound systems with separate monitor consoles use transformer isolated splitters to send the signals to two places, but these are prohibitively expensive for most working bands and small clubs. Y-cables can be used to split microphone signals, but they can get messy and are somewhat unreliable.

Enter the P4M Personal Monitor Mixer. Developed by Shure, the P4M combines a four-channel mixer with a splitter to pass the input signals on to the FOH or monitor mixers, or to other P4M mixers. Splits are accomplished inside the mixer, reducing cable clutter on stage. The mixer can be placed near the performer for easy access to the controls. For the non-technical musician, the P4M is very easy to control – just volume and pan. Inputs are combination XLR-1/4" phone jacks that accept microphone or line-level signals. The performer can choose which four inputs are most important for them to have control of, and take them directly into the P4M. These signals will pass directly to four XLR outputs on the back of the P4M, completely unaffected by the mixer's controls. The Mix Outputs of the P4M are then input directly to the personal monitor system.

Obviously, many performers will need to hear more than just four things, so the P4M also includes a set of stereo auxiliary inputs to accept an additional mix from another console. The auxiliary inputs can also be used to connect multiple P4Ms together to expand the number of inputs. Connecting the mix outputs of one P4M to the Aux Inputs of another will now get the performer a total of eight inputs.



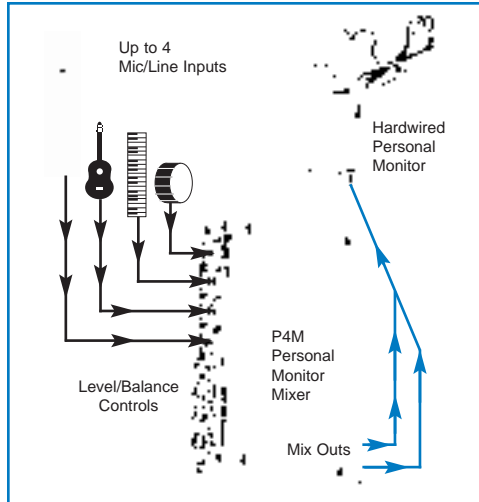
P4M front panel



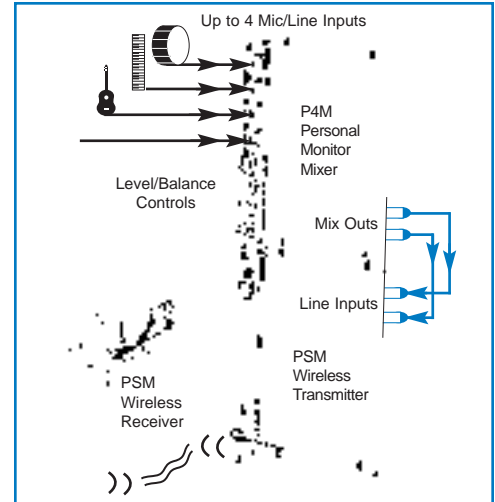
P4M back panel

The split outputs of the P4M can be used to cascade the input signals to multiple P4M mixers. For example, each member of a typical wedding band has their own P4M, and they all want individual control of the lead singers microphone level. The lead vocal microphone is plugged into the first P4M, then the split output of that microphone is connected to the next P4M, and so on until each band member has the lead singer's microphone in their monitor mix. The split output of the last P4M is then connected the FOH mixer. Based on subjective listening tests, a microphone can be routed through three to four P4Ms before any degradation of sound quality occurs. This loss may vary from microphone to microphone, so experimentation may be necessary to find the setup that will work the best. If a signal needs to route through more than four P4Ms, use a line level signal from the mixer instead of a microphone. When there is a limited number of mixes available from the main mixer, a submix can be created and routed through the P4M inputs.

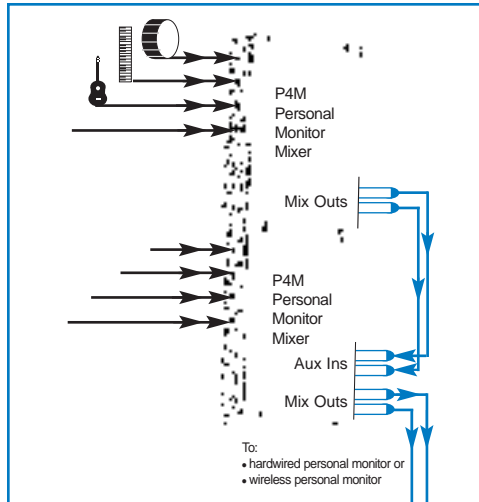
The P4M is extremely flexible, and the ways in which it can be implemented are almost limitless. Following are a few examples of personal monitor systems that use the P4M, but this is by no means an exhaustive list. If you can think of it, there is probably a way to do it.



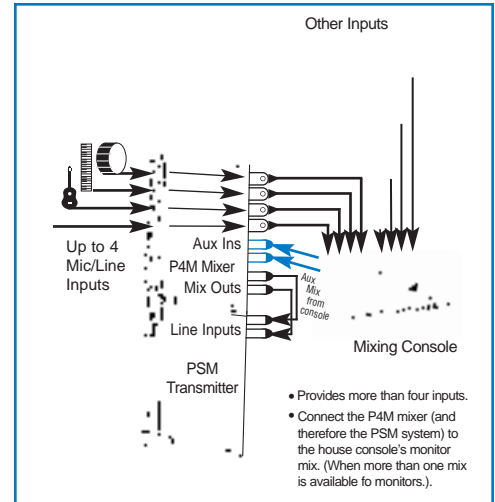
Connecting the P4M to a Hardwired Personal Monitor



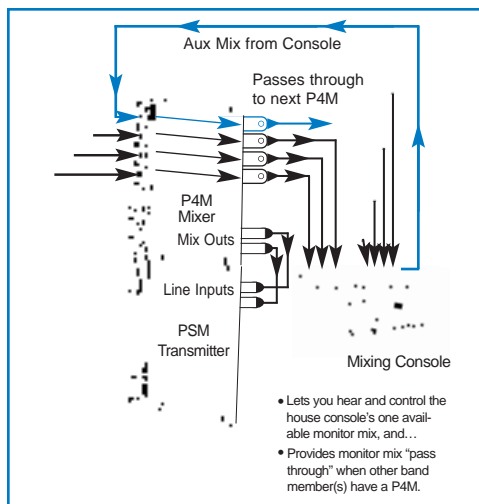
Connecting the P4M Mixer to a Wireless Personal Monitor



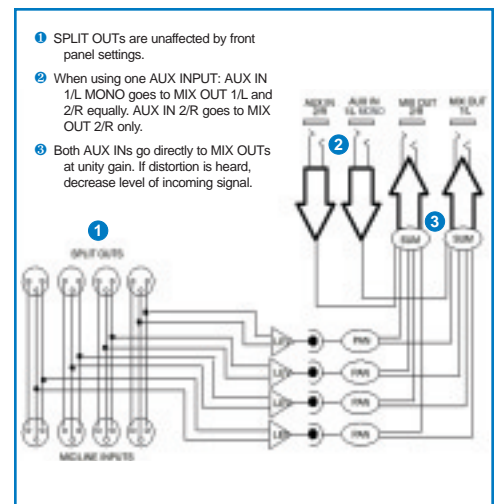
Connecting the P4M to another P4M (which provides more than four outputs)



Connecting the P4M to the House Console



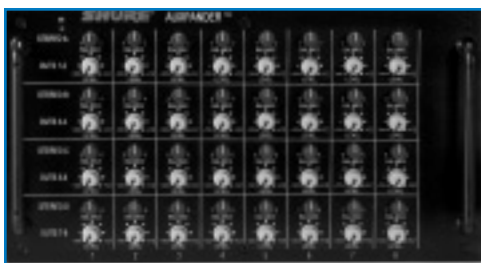
Connecting your P4M to the House Console (when only one mix is available from the console)



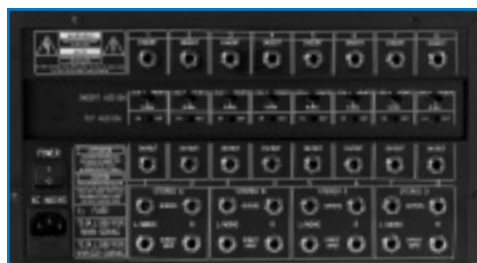
P4M Block Diagram

“I could not wait to get my hands on the new Shure PSM 600 in-ear monitor system, after I had tried out the prototype in 1996. When it finally arrived in 1997 on The Who’s Quadrophenia Tour, I was not disappointed. It was even better than I remembered. The clarity is fantastic and the signal-to-noise ratio is amazing. I also find that the limiter that is incorporated into the system is excellent. Lately, I have been using it in the studio for drummers and find it very useful. Thus proving that, as always, Shure products are great in the studio and on the road.”

—Bob Pridden,
Sound Designer
for The Who



AuxPander front panel



AuxPander back panel

AuxPander

With so many processors available to musicians and audio engineers, occasionally the number of auxiliary sends and returns can equal or exceed the number of signals present in a mix. The routing capabilities of most common mixing consoles are adequate for simple interconnect tasks, but can be severely taxed when required to provide individual monitoring or multi-track recording output busses. In today's audio world, mixing devices must be able to route any, many or all inputs to any, many or all outputs - each at different relative levels. Large frame mixing consoles provide this capability with a featured master section known as the matrix. These consoles tend to be significantly expensive, large, and heavy. Furthermore, the learning curve for these mixers is quite steep. AuxPander (a contraction of the words Auxiliary and Expander) is a matrix that seamlessly interconnects with virtually any mixing console. Adding an AuxPander to even the smallest mixing console greatly increases its routing and mixing capabilities.

General Description

The AuxPander is an eight input/eight output (8 X 8), line level, matrix mixer that derives its inputs from the insert points of a standard mixing console. These insert points may be from any combination of input channels or subgroups, allowing for a number of connection possibilities limited only by the user's imagination.

Once inserted, AuxPander offers up to four stereo or eight mono, prefader, auxiliary sends. These sends can be used to create mixes for a variety of applications, including personal or loudspeaker monitor mixes, multitrack recording and monitoring, effect sends (in stereo!), zone mixing and mix-minus matrixing.

Other devices that would normally use the inserts, such as compressors or gates, can be connected directly to the AuxPander's own insert points. Therefore, adding an AuxPander does not compromise the ability to implement the normal variety of audio processors. Moreover, AuxPander's insert points feature the ability to assign the inserted processor to either mixer or both, if necessary.

Using AuxPander with Personal Monitors

Many performers who use personal monitors require individualized mixes. While sharing monitor mixes is perfectly acceptable for some, others desire a completely custom mix. Although multiple mixes can easily be achieved with personal monitor systems, the limiting factor for the number of available mixes is usually the mixing console. A typical console will have four auxiliary sends. Usually, two sends are dedicated to effects, leaving two sends for monitors. What if your band has four members who each need their own mix? Even if you dropped the effects and used those sends for monitors, if they are not switchable to prefader, then the band will hear the sound engineer's fader moves in their ears. Also, if stereo mixes are required, the number of necessary auxiliary sends doubles.

“My floor-wedge spent the rest of the tour in the back of the truck. It is such a relief to finally hear all the nuances and detail of my efforts, particularly in the case of the flute. Now after 29 years of playing it, I can finally hear every note! Well done! Shure has come up with another winner.”

—Ian Anderson

The primary use for an AuxPander is to provide additional monitor mixes, in either a live performance or recording environment. Since more and more performers are demanding their own individual mix, AuxPander allows for the creation of up to four stereo or eight mono individual monitor mixes, without affecting the functionality of the main console. Combinations of stereo and mono mixes can also be generated. For example, one stereo and six mono, two stereo and four mono, or three stereo and two mono mixes can be prepared from a single AuxPander. This flexibility is useful when performers are using combinations of stereo personal and mono loudspeaker monitor systems.

Using AuxPander with Loudspeaker Monitors

Even though some performers continue to monitor through wedge loudspeakers, they still desire their own personalized mix in their wedge. Since many mixing consoles do not provide enough dedicated outputs for each performer to have their own mix, compromises have to be made. Most commonly, two or more performers have to share a mix. Rarely do two performers wish to hear exactly the same signals in their monitor at exactly the same relative levels. Even though they may have their own speaker cabinet, they may not have a dedicated mix output to hear exactly what they want to hear in it. Just as with personal monitors, AuxPander provides separate mixes for each member regardless of individual requirements.

For performing groups who use a combination of personal and loudspeaker monitors, an AuxPander is almost essential. While it is possible to daisy-chain a mix from a PSM transmitter's Loop Out jack to a power amplifier and loudspeaker, rarely will a mix created for in-ear monitors sound good through a loudspeaker. Accommodations need to be made for the acoustic environment, that are completely independent from that of a sealed ear canal. Unlike personal monitors, loudspeaker monitors contribute sound

pressure level to the acoustic space in which they are heard. Hence, gain-before-feedback (See Chapter 2) can become an issue. A vast array of outboard equipment, including equalizers, gates, and compressors, is typically employed in an attempt to tame an unfriendly acoustic environment. While necessary for loudspeaker monitoring, these devices could be distracting to someone using personal monitors.

Other Applications for AuxPander

The usage of AuxPander is not limited to the live performance environment. Recording studios commonly need to provide headphone mixes to many musicians. AuxPander can expand the number of independent mixes beyond that of the typical mixer. AuxPander also makes a good front-end for multi-track recording, especially with the proliferation of MDM, or modular digital multi-track recorders in use today. Particularly for live concert recording, where the engineer would prefer to have individual level control to the recorder, the AuxPander gives that functionality without taking away from the needs of the live sound engineer. The variety of applications for AuxPander is beyond the scope of this publication.

Setting Up an AuxPander

Let's take a look at a typical AuxPander application. Assume the FOH console has four auxiliary sends, all of which are used for effect sends. Rather than carry a monitor board and splitter, the band decided to use an AuxPander for monitor sends. The following inputs are required for the band's personal monitor system:

- Kick Drum
- Snare Drum
- Rack Tom
- Floor Tom
- Bass Guitar
- Electric Guitar
- Lead Vocal
- Background Vocal

“We use a wedge monitor system, with the exception of Ed (Kowalczyk), who is on the Shure PSM. He thinks it makes him sing better. Plus it gives us a cleaner stage, and lower stage volume. I think all the front of house guys will love it.”

—Jon Lemmon,
Front of House
Engineer for Live

First, connect the main mixer's inserts to the In/Out jacks of the AuxPander. Second, connect AuxPander's output jacks to the inputs of the monitor system. Assuming each performer has their own transmitter, and is monitoring in stereo, then the connections are simple: Left Out 1 and Right Out 1 from AuxPander to Left and Right inputs of the first transmitter, and so on until all four mixes are wired.

Now assume that the FOH engineer wants to insert gates on the drums, and compressors on the vocals and bass guitar. The gates should only affect the main mix, not the monitor sends. The vocals need to route through a reverb processor that doesn't affect the FOH mix. The bass compressor should be present at both places. First, all these devices are accessed from AuxPander's insert points. Then, use the Insert Assign switch to choose which mixer the insert point affects. Set the channels with gates to Remote, which will insert the gates only at the main console. The vocal channels should be set to Local, which will return the reverb only to AuxPander. Set the bass channel to Global, which will compress the signal at both mixers.

AuxPander also includes a Tip Assign switch. The insert points on most mixers usually send on the Tip and return on the Ring. Some manufactures (Soundcraft, Ramsa, and a few others) use the opposite: Ring send and Tip return. To accommodate both types of mixer, use the AuxPander's Tip Assign switch.

Lastly, the Direct In jacks allow for insertion of signals directly to a given set of outputs. This feature is useful for sending signals into AuxPander that do not need to return to the main mixer. For example, a mixer with direct outputs on its subgroups could send a mixed signal from that subgroup directly into AuxPander without using any additional channel inserts. If the lead vocalist in the above example needed effects in his mix, all the effect returns could be bussed to a subgroup, and the subgroup output could be connect to AuxPander's Direct In jack.

AuxPander and the P4M can be used in tandem for even more possibilities. The mind boggles...

“After screaming my guts out for 15 years, my dream has finally come true – perfect monitors every night. No more hand over the ear to hear yourself. No more wedges between me and the crowd. No more straining. No more screaming. Just clear, perfect sound, night after night. Which is what it's all about. Thank you, Shure.”

–Sebastian Bach

Mixing

One of the advantages of having a professional sound engineer or monitor engineer is years of experience in mixing sound. This skill cannot be learned overnight. For bands that are new to personal monitors, there is strong temptation to try to create a CD quality mix for in-ears. While this is certainly possible with a trained sound engineer and the right equipment, it is unlikely that someone unfamiliar with the basic concepts behind mixing will be able to successfully imitate a professional mix. Does this mean you shouldn't use personal monitors without a monitor tech? Certainly not!

A common mistake made by in-ear monitor novices is to put everything but the kitchen sink into the mix.

Here's an alternative to the “everything-in-the-mix” method:

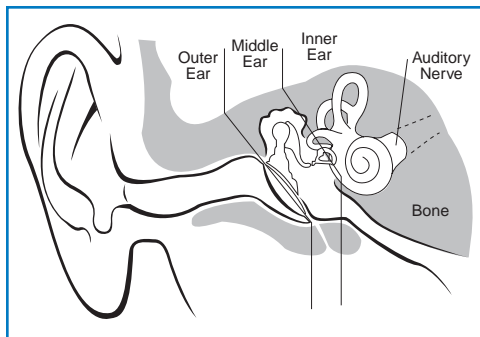
1. Put the monitors in your ears, and turn the system on. DO NOT put any instruments in your mix yet!
2. Try to play a song. While you are playing, determine what you need to hear more of.
3. Begin bringing instruments into the mix, one at a time. Chances are, you will need vocals first, since those are often the only unamplified “instruments” on stage.
4. Only turn things up as loud as necessary, and resist the temptation to add instruments to the mix that you can hear acoustically.

CHAPTER 7

HEARING, THE DB, AND SPL

No discussion of monitoring systems would be complete without some discussion of human hearing. The brain's ability to interpret the vibration of air molecules as sound is not entirely understood, but we do know quite a bit about how the ear converts sound waves into neural impulses that are understood by the brain.

The ear is divided into three sections; the outer, middle, and inner ear. The outer ear serves two functions – to collect sound and act as initial frequency response shaping. The outer ear also contains the only visible portion of the hearing system, the pinna. The pinna is crucial to localizing sound. The ear canal is the other component of the outer ear, and provides additional frequency response alteration. The resonance of the ear canal occurs at approximately 3 kHz, which, coincidentally, is right where most consonant sounds exist. This resonance increases our ability to recognize speech and communicate more effectively. The middle ear consists of the ear drum and the middle ear bones (ossicles). This section acts as an impedance-matching amplifier for our hearing system, coupling the relatively low impedance of air to the high impedance of the inner ear fluids. The ear drum works in a similar manner to the diaphragm of a microphone, it moves in sympathy to incoming sound waves, and transfers those vibration to the ossicles. The last of these bones, the stapes (or stirrup, as our grade school teachers called it), strikes an oval-shaped window that leads to the cochlea, the start of the inner ear. The cochlea

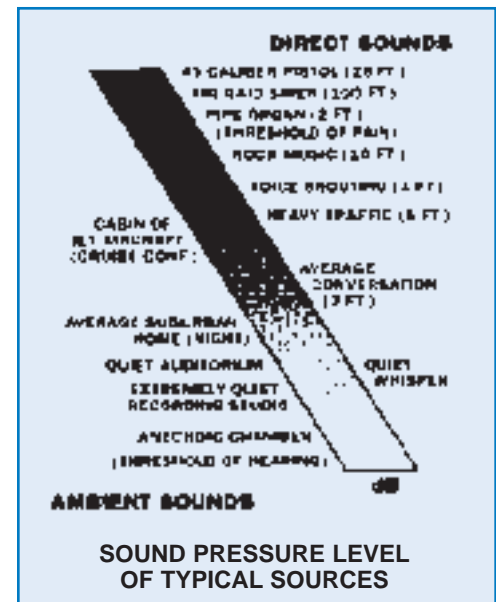


Ear Anatomy

contains 15,000 to 25,000 tiny hairs, known as cilia, which bend as vibrations disturb the fluids of the inner ear. This bending of the cilia sends neural impulses to the brain via the auditory nerve, which the brain interprets as sound.

Hearing loss occurs as the cilia die. Cilia begin to die from the moment we are born, and they do not regenerate. The cilia that are most sensitive to high frequencies are also the most susceptible to premature damage. Three significant threats to cilia are infection, drugs, and noise. Hearing damage can occur at levels as low as 90 dB SPL. (See sidebar about the decibel.) According to OSHA (Occupational Safety Health Administration), exposure to levels of 90 dB SPL for a period of 8 hours could result in some damage. Of course, higher levels reduce the amount of time before damage occurs.

- 90 dB SPL – 8 Hours
- 95 dB SPL – 4 Hours
- 100 dB SPL – 2 Hours
- 105 dB SPL – 1 Hour
- 110 dB SPL – 30 Minutes
- 115 dB SPL – 15 Minutes



Hearing conservation is important to everyone in the audio industry. As mentioned before, an in-ear monitor system can assist in helping to prevent hearing damage – but they are not foolproof protection. The responsibility for safe hearing is now in the hands of the performer. At this point, there are no direct correlations between where to set your volume control and how much SPL is present in your

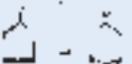
ears. Here are a few suggestions, though, to help protect your hearing.

1. **Use the limiter!** All Shure PSM systems feature a built-in limiter. We strongly recommend using it at all times. The limiter will prevent the sound level from exceeding a preset threshold, which protects your ears from sudden loud bursts or unexpected, high-level transients.
2. **If it hurts, it's too loud.**
3. **Temporary Threshold Shift** – Pay attention to what your ears are telling you. TTS is characterized by a “stuffiness,” or compressed feeling, like someone stuck cotton in your ears. Ringing (or “tinnitus”) is another symptom of TTS, which is the ear’s way of telling you that you are being exposed to sound levels that are too extreme. Please note, though, that hearing damage may have occurred even if you do not experience ringing. In fact, the majority of people who have hearing damage never reported any ringing. After experiencing TTS, most of your hearing will recover. Permanent damage has possibly occurred, though. The effects of TTS are cumulative, so if you regularly experience the above effects, your monitoring level is too loud and hearing damage will occur with repeated exposure to those levels. Turn it down!

House Ear Institute Hotline: (213) 483-4431 Website: www.hei.org	H.E.A.R. Hotline: (415) 409-3277 Website: www.hearnet.com
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Sensaphonics Hearing Conservation
660 N. Milwaukee Avenue, Chicago IL 60622
Toll Free: (877) 848-1714 In Chicago: (312) 432-1714
Fax: (312) 432-1738
Website: www.sensaphonics.com
E-mail: saveyourears@sensaphonics.com

The beauty of personal monitors is the fact that *you* can turn it down. Not only do personal monitors offer improved sound quality and convenience, but they put you back in control of your sound.

1. Compare	2. Compress	3. Scale (x 20)
	$10^0=1$	0
	$10^1=10$	20
	$10^2=100$	40
	$10^3=1000$	60
	$10^4=10,000$	80
	$10^5=100,000$	100
	$10^6=1,000,000$	120
b		
a		
b/a		

Decibel Scale for dBV or dB SPL

The Decibel

The decibel (dB) is an expression often used in electrical and acoustic measurements. The decibel is a number that represents a ratio of two values of a quantity such as voltage. It is actually a logarithmic ratio whose main purpose is to scale a large measurement range down to a much smaller and more useable range. The form of the decibel relationship for voltage is:

$$\text{dB} = 20 \times \log(V1/V2)$$

where 20 is a constant, V1 is one voltage, V2 is the other voltage, and log is logarithm base 10.

Examples:

What is the relationship in decibels between 100 volts and 1 volt:

$$\begin{aligned}\text{dB} &= 20 \times \log(100/1) \\ \text{dB} &= 20 \times \log(100) \\ \text{dB} &= 20 \times 2 \quad (\text{the log of 100 is 2}) \\ \text{dB} &= 40\end{aligned}$$

That is, 100 volts is 40dB greater than 1 volt.

What is the relationship in decibels between 0.001 volt and 1 volt?

$$\begin{aligned}\text{dB} &= 20 \times \log(0.001/1) \\ \text{dB} &= 20 \log(0.001) \\ \text{dB} &= 20 \times (-3) \quad (\text{the log of 0.001 is -3}) \\ \text{dB} &= -60\end{aligned}$$

That is, 0.001 volt is 60dB less than 1 volt.

Similarly:

- if one voltage is equal to the other they are 0dB different;
- if one voltage is twice the other they are 6dB different;
- if one voltage is ten times the other they are 20dB different.

Since the decibel is a ratio of two values, there must be an explicit or implicit reference value for any measurement given in dB. This is usually indicated by a suffix on the decibel value such as: dBV (reference to 0.0002 microbar which is 0dB Sound Pressure Level).

One reason that the decibel is so useful in certain audio measurements is that this scaling function closely approximates the behavior of human hearing sensitivity. For example, a change of 1dB SPL is about the smallest difference in loudness that can be perceived while a 3dB SPL change is generally noticeable. A 6dB SPL change is quite noticeable and finally, a 10dB SPL change is perceived as “twice as loud.”

Radio Transmission

Radio refers to a class of time-varying electromagnetic fields created by varying voltages and/or currents in certain physical sources. These sources may be “artificial,” such as electrical power and electronic circuits, or “natural,” such as the atmosphere (lightning) and stars (sunspots). The electromagnetic field variations radiate outward from the source forming a pattern called a radio wave. Thus, a radio wave is a series of electromagnetic field variations travelling through space. Although, technically, any varying source of voltage or current produces a varying field near the source, here the term “radio wave” describes field variations that propagate a significant distance from the source.

A sound wave has only a single “field” component (air pressure). Variations in this component create a pattern of air pressure changes along the direction that the sound wave travels but otherwise have no particular orientation. In contrast, a radio wave includes both an electric field component and a magnetic field component. The variations in these components have the same relative pattern along the direction that the radio wave travels but they are oriented at a 90 degree angle to each other as illustrated in the accompanying figure. In particular, it is the orientation of the electric field component which determines the angle of “polarization” of the radio wave. This becomes especially important to the design and operation of antennas.



Figure 1: Radio Wave

Like sound, a radio wave can be described by its frequency and its amplitude. The frequency of a radio wave is the time rate of the field variations measured in Hertz (Hz), where 1 Hz equals 1 cycle-per-second. The radio spectrum, or range of frequencies, extends from a few Hertz through the Kilohertz (KHz) and Megahertz (MHz) ranges, to beyond the Gigahertz (GHz) range. The suffixes KHz, MHz, and GHz refer to thousands, millions, and billions of cycles-per-second respectively. As far as is presently known, humans are directly sensitive to radio waves only at frequencies in the range of a few million GHz, which are

perceived as visible light, and at those frequencies in the range just below visible light, which are perceived as heat (infrared radiation). The overall radio spectrum includes both natural and artificial sources as indicated by Figure 2.

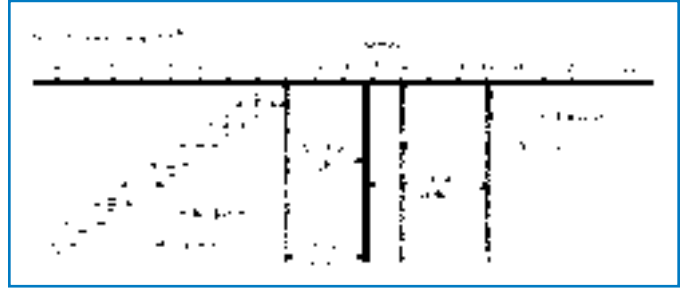


Figure 2: Radio Frequency Spectrum

The amplitude of a radio wave is the magnitude of the field variations and is the characteristic that determines the “strength” of the radio wave. Specifically, it is defined to be the amplitude of the electric field variation. It is measured in volts per unit length and ranges from nanovolts/meter (nV/m) to kilovolts/meter (KV/m), where nV refers to one billionth of a volt and KV refers to one thousand volts. The minimum level required for pickup by a typical radio receiver is only a few tens of microvolts (uV, a millionth of a volt) but much higher levels can be found near transmitters and other sources. The wide range of radio wave amplitudes that may be encountered in typical applications requires great care in the design and use of wireless systems, particularly receivers.

Another characteristic of radio waves, related to frequency, is wavelength. The wavelength is the physical distance between the start of one cycle and the start of the next cycle as the wave moves through space. Wavelength is related to frequency by the speed at which the radio wave travels.

The speed of radio waves (in a vacuum) is equal to approximately 3×10^8 meter/sec, or about 186,000 miles/sec, the same as the speed of light. It does not change with frequency or wavelength but is related to them in the following way: the frequency of a radio wave, multiplied by its wavelength always equals the speed of light. Thus, the higher the radio frequency, the shorter the wavelength, and the lower the frequency, the longer the wavelength. Typical wavelengths for certain radio frequencies are given in Figure 3. Wavelength also has important consequences in the design and use of wireless systems, particularly for antennas.

Unlike sound, radio waves do not require a physical substance (such as air) for transmission. In fact, they “propagate” or travel most efficiently through the vacuum



Figure 3: Radio Wavelength Chart

of space. However, the speed of radio waves is somewhat slower when travelling through a medium other than vacuum. For example, visible light travels more slowly through glass than through air. This effect accounts for the “refraction” or bending of light by a lens. Radio waves can also be affected by the size and composition of objects in their path. In particular, they can be reflected by metal if the size of the metal object is comparable to or greater than the wavelength of the radio wave. Large surfaces can

reflect both low frequency (long wavelength) and high frequency (short wavelength) waves, but small surfaces can reflect only high frequency (short) radio waves.

Interestingly, a reflecting metal object can be porous, i.e. it can have holes or spaces in it. As long as the holes are much smaller than the wavelength, the metal surface will behave as if it were solid. This means that screens, grids, bars, or other metal arrays can reflect radio waves whose wavelength is greater than the space between the array elements and less than the overall array size. If the space between elements is larger than the wavelength, the radio waves will pass through the array. The metal screen on the glass door of a microwave oven reflects microwaves back into the oven but allows (shorter) light waves to pass through so that the inside is visible.

Even metal objects which are smaller than the wavelength are able to bend or “diffract” radio waves. Generally, the size, location, and quantity of metal in the vicinity of radio waves will have significant effect on their behavior.

Non-metallic substances (including air) do not reflect radio waves but are not completely transparent either. To some degree, they generally “attenuate” or cause a loss in the strength of radio waves that pass through them. The amount of attenuation or loss is a function of the thickness and composition of the material and also a function of the radio wavelength. In practice, dense materials produce more losses than lighter materials and long radio waves (low frequencies) can propagate greater distances through “lossy” materials than short radio waves (high frequencies). The human body causes significant losses to short radio waves passing through it.

An object which is large enough to reflect radio waves or dense enough to attenuate them can create a “shadow” in the path of the waves which can greatly hamper reception of radio in the area beyond the object.



Figure 4: Radio wave propagation vs. wavelength when encountering conductive obstacles

A final parallel between sound waves and radio waves lies in the nature of the radio wave pattern or “field” produced by various sources at a given location. If reflections are present (which is nearly always the case indoors), the radio field will include both direct waves (those that travel by the shortest path from the source to the location) and indirect waves (those that are reflected). Radio waves, like sound waves, become weaker as they travel away from their source, at a rate governed by the inverse-square law: at twice the distance, the strength is decreased by a factor of four (the square of two). The radio waves that arrive at a given location, by direct or indirect paths, have different amplitudes related to the strength of the original source(s) and the amount of loss due to reflections, material attenuation and the total distance traveled.

After many reflections radio waves become weaker and essentially non-directional. They ultimately contribute to ambient radio “noise,” that is, general radio energy produced by many natural and man-made sources across a wide range of frequencies. The strength of ambient radio noise is relatively constant in a given area, that is, it does not diminish with distance. The total radio field at a given location consists of direct waves, indirect waves and radio noise.

Radio noise is nearly always considered to be undesirable. The direct and indirect waves may come from both the desired source (the intended transmission) and undesirable sources (other transmissions and general radio energy emitters). Successful radio reception depends on a favorable level of the desired transmission compared to undesirable transmissions and noise.

This discussion of radio transmission has so far dealt only with the basic radio wave. It is also necessary to consider how information is carried by these waves. Audio “information” is transmitted by sound waves which consist of air pressure variations over a large range of amplitudes and frequencies. This combination of varying amplitudes and varying frequencies creates a highly complex sound field. These varying pressure waves are able to be processed directly by our auditory systems to perceive speech, music, and other intelligible sounds (information).

Radio “information” is generally transmitted using only one frequency. This single electromagnetic wave is varied in amplitude, frequency, or some other characteristic (such as phase) and for most radio transmissions neither the wave nor its variation can be detected or processed directly by human senses. In fact, the wave itself is not the information but rather the “carrier” of the information. The information is actually contained in the amplitude variation or frequency variation, for example. When a radio wave contains information it is called a radio “signal.” The term for variation of radio waves is “modulation.” If the amplitude of the wave is varied the technique is called Amplitude Modulation or AM. If the frequency is varied, it is called Frequency Modulation or FM.

The amount of information that can be carried in a radio signal depends on the amount and type of modulation that can be applied to the basic radio wave as well as the base frequency of radio wave. This is limited by physics to some extent, but is also limited by regulatory agencies such as the FCC. For AM signals, the radio wave has a single (constant) frequency of some basic amplitude (determined by the transmitter power). This amplitude is varied up and down (modulated) by the audio signal to create the corresponding radio signal. The maximum (legal) amount of amplitude modulation allows an audio signal of only limited frequency response (about 50-9000 Hz) and limited dynamic range (about 50 dB).

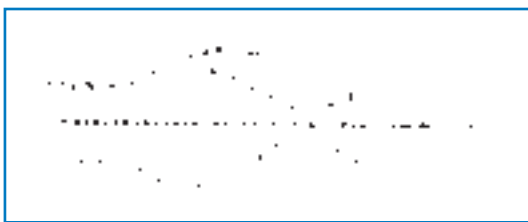


Figure 5: Modulation AM Carrier

For FM signals, the radio wave has a constant amplitude (again determined by transmitter power) and a basic frequency. The basic radio frequency is varied up and down

(modulated) by the audio signal to create the corresponding radio signal. This frequency modulation is called “deviation” since it causes the carrier to deviate up and down from its basic or unmodulated frequency.

The deviation is a function of the amplitude of the audio signal and is usually measured in kilohertz (KHz). Typical values of deviation in wireless microphone systems range from about 12KHz to 45KHz depending on the operating frequency band. The maximum (legal) amount of deviation allows an audio signal of greater frequency response (about 50-15,000 Hz) and greater dynamic range (more than 90 dB) than does AM.

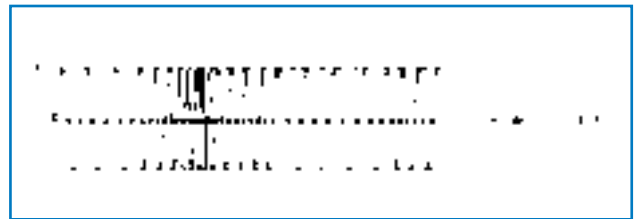


Figure 6: Modulation FM Carrier

It should be noted here that all of the systems discussed in this presentation use the FM technique. The reasons for this are the same as are apparent in commercial broadcast systems. More “information” can be sent in the typical FM signal, allowing higher fidelity audio signals to be transmitted. In addition, FM receivers are inherently less sensitive to many common sources of radio noise, such as lightning and electrical power equipment, because the AM component of such interference is rejected.

Antennas

In addition to the circuitry contained inside transmitters and receivers, a critical circuitry element is often located outside the unit: the antenna. In fact, the design and implementation of antennas is at least as important as the devices to which they are attached. Although there are certain practical differences between transmitting and receiving antennas there are some considerations that apply to both. In particular, the size of antennas is directly proportional to wavelength (and inversely proportional to frequency). Lower radio frequencies require larger antennas, while higher frequencies use smaller antennas.

Another characteristic of antennas is their relative efficiency at converting electrical power into radiated power and vice versa. An increase of 6 dB in radiated power, or an increase of 6 dB in received signal strength can correspond to a 50% increase in range. Likewise, a loss of 6 dB in signal may

result in 50% decrease in range. Though these are best (and worst) case predictions, the trend is clear: greater antenna efficiency can give greater range.

The function of an antenna is to act as the interface between the internal circuitry of the transmitter (or receiver) and the external radio signal. In the case of the transmitter, it must radiate the desired signal as efficiently as possible, that is, at the desired strength and in the desired direction. Since the output power of most transmitters is limited by regulatory agencies to some maximum level, antenna efficiency is critical.

The smallest simple antenna that is consistent with reasonable transmitter output is an antenna which is physically (and electrically) one quarter as long as the wavelength of the radio wave frequency being transmitted. This is called a “1/4 wave” antenna.

For transmitter applications requiring even smaller antenna size a short “rubber duckie” antenna is sometimes used. This type is still (electrically) a 1/4 wave antenna, but it is wound in a helical coil to yield a shorter package. There is some loss in efficiency due the smaller “aperture” or physical length. In addition, these antennas have a narrower bandwidth.

The radio wave pattern emitted by the 1/4 wave antenna is omnidirectional in the plane perpendicular to the axis of the antenna. For a vertically oriented 1/4 wave antenna the radiation pattern is omnidirectional in the horizontal plane. There is very little output along the axis of the antenna. A three-dimensional representation of the field strength from a vertical antenna would resemble a horizontal doughnut shape with the antenna passing through the center of the hole.

Recall that a radio wave has both an electric field component and a magnetic field component. A vertically oriented 1/4 wave transmitter antenna radiates an electric field component that is also vertical (while the magnetic field component is horizontal). This is said to be a “vertically polarized” wave. Horizontal orientation of the antenna produces a “horizontally polarized” wave.

Unidirectional antennas are also available for wireless systems. These designs are comprised of a horizontal boom with multiple transverse elements and are of the same general type as long range antennas for television reception. They can achieve high gain (up to 10 dB compared to the 1/4 wave type) in one direction and reduce multipath drop outs.

Two common types are the Yagi and the log-periodic. The Yagi consists of a dipole element and one or more additional elements: those located at the rear of the boom are larger than the dipole element and reflect the signal back to the dipole while those located at the front are smaller than the dipole and act to direct the signal on to the dipole. The Yagi has excellent directivity but has a fairly narrow bandwidth and is usually tuned to cover just one TV channel (6 MHz).

The log-periodic achieves greater bandwidth than the Yagi by using multiple dipole elements in its array. The size and spacing between the dipoles varies in a logarithmic progression so that at any given frequency one or more dipoles are active while the others are functioning as reflecting or directing elements, depending on their size and location relative to the active element(s). The longer the boom and the greater the number of elements the greater is the bandwidth and the directivity. Although these directional antennas are somewhat large (3-5 ft. wide for VHF) and may be mechanically cumbersome to mount, they can provide increased range.

In receiver applications, the antenna must pick up the desired radio signal as efficiently as possible. Since the strength of the received signal is always far less than that of the transmitted signal this requires that the antenna be very sensitive to the desired signal and in the desired direction.

Again, the minimum size for adequate reception is 1/4 wavelength. A whip or telescoping antenna of this size is supplied with most receivers, and it too is omnidirectional in the horizontal plane when it is vertically oriented. An important consideration in the performance of a 1/4 wave receiving antenna is that its efficiency depends to some extent on the presence of a “ground plane,” that is, a metal surface at least 1/4 wave long in one or both dimensions and electrically connected to the receiver ground at the base of the antenna. Typically, the receiver chassis or receiver PC board to which the antenna is attached acts as a sufficient ground plane.

If more sensitivity is desired, or if it is necessary to mount an omnidirectional antenna remotely from the receiver, 1/2 wave or 5/8 wave antennas are often used. These antennas have a theoretical “gain” (increase of sensitivity) up to 3 dB greater than the 1/4 wave antenna in some configurations. This can translate into increased range for the system. However, the 5/8 wave antenna, like the 1/4 wave type, only achieves its performance with an appropriate ground plane. Without a ground plane unpredictable effects may occur resulting in asymmetric pickup patterns and potential signal loss due to the non-ideal cable/antenna interface.

A properly designed 1/2 wave antenna does not require a ground plane, allowing it to be remotely mounted with relative ease. It can also maintain proper impedance at the cable/antenna interface or can be directly attached to a receiver or antenna distribution system. In addition, it is resistant to the effects of electrical noise which might otherwise be picked up at the interface.

When antenna size is an issue, such as for portable receivers, the previously mentioned 1/4 wave rubber duckie is an option. UHF designs can use 1/2 wave rubber duckies because of the shorter wavelengths. Another relatively small size remote antenna can be found in the form of a 1/4 wave antenna with an attached array of radial elements that function as an integral ground plane. Both of these types are omnidirectional in the horizontal plane when mounted vertically.

For maximum efficiency receiving antennas should be oriented in the same direction as the transmitting antenna. In the same way that a transmitter antenna produces a radio wave that is “polarized” in the direction of its orientation, a receiver antenna is most sensitive to radio waves that are polarized in its direction of orientation. For example, the receiving antenna should be vertical if the transmitting antenna is vertical.

Range of Wireless Systems

A logical question concerning wireless performance is the transmission range of various systems. Unfortunately, the answer is much more complicated than a simple distance measurement. Ultimately, the receiver must be able to pick up a “useable” signal from the transmitter. “Useable” means that the strength of the desired signal is within the sensitivity range of the receiver and further that it is sufficiently stronger than (or different from) undesirable signals and RF noise to produce an acceptable signal-to-noise ratio at the audio output of the receiver. Elements that affect useability are the transmitter/antenna, the transmission path, the receiver/antenna and RFI. Some characteristics of these elements are controllable, some are not.

Important transmitter characteristics are power output and antenna efficiency. Maximum power is limited by government regulations and battery capability. Antenna efficiency is limited by size and design. Recall that the efficiency of typical wireless transmitter antennas is fairly low, about 10% or less for VHF. This means that for a 50 mW VHF transmitter the effective radiated power (ERP) is less than 5 mW. This may be further attenuated by proximity to the body or other lossy objects.

Important receiver characteristics are antenna efficiency, receiver sensitivity and the ability of the receiver to reject unwanted signals and noise. Antenna efficiency is again limited by size and design but receiver antennas tend to be much more efficient than transmitter antennas since they can be made large enough to be better tuned to the proper frequency. Other receiver characteristics are limited by design. Both elements are limited by cost.

The transmission path is characterized by distance, intervening obstructions and propagation effects. Losses due to these characteristics are generally frequency dependent: the higher the frequency the greater the loss. Once the operating frequency is chosen, only the path length and antenna locations are controllable. These are usually limited by the application itself. Under good conditions (line-of-sight) at a distance of about 100 ft. the field strength of the signal from a 50 mW transmitter is on the order of 1000 uV/m, well within the range of sensitivity of a typical receiver.

Finally, RFI is characterized by its spectrum, that is, its distribution of amplitude and frequency. It typically consists of both broadband noise and discrete frequencies. However, its strength can be comparable to or greater than the desired signal in poor conditions. Except for a few predictable sources it is largely uncontrollable.

Rather than quote a specific maximum operating distance most manufacturers of wireless systems (microphone or personal monitor) give a “typical” range. For systems of the type discussed here (50 or 100 mW, VHF or UHF) the typical range may vary from 100 ft. to 1000 ft. The lower number represents a moderately severe environment while the upper figure might be achieved in absolute ideal conditions. Extremely poor conditions could result in a range of only 50 feet or less. It is impossible to accurately predict the range of an arbitrary wireless microphone system in an arbitrary application.

Crystal Controlled vs. Frequency Synthesis

Crystal controlled wireless units can be designed with wide audio frequency response, low noise, low distortion, and relatively long battery life. They are the most cost effective choice for fixed frequency applications involving a moderate number of simultaneous systems. One limitation inherent to a crystal controlled transmitter is the generation of spurious emissions due to output multiplier stages, though these can generally be kept to a minimum with careful design.

For tuneable systems, frequency synthesis is the most practical technique. The absence of spurious emissions from the transmitter also simplifies coordination of multiple

systems. However, it is more difficult (and more expensive) to design equally low noise, low distortion frequency synthesized systems. A limitation inherent to the audio frequency response of this type of transmitter results from the use of a sharp lo-cut filter to prevent very low audio frequencies from interfering with the PLL control circuit. This places a lower limit on the audio frequency range that may be transmitted. Special techniques are required to achieve extended low frequency response in frequency synthesized systems. In addition, the greater power consumption of frequency synthesized transmitters reduces battery operating time.

For most fixed applications crystal controlled systems are suitable. Frequency synthesized systems should be considered when frequency agility is a primary requirement or if there are other included features that are desirable for the application.

Above excerpted from Selection and Operation of Wireless Microphone Systems by Tim Vear (AL1247A).

Frequency Compatibility

A commonly overlooked and misunderstood concept in wireless technology is the issue of frequency compatibility. The frequencies that wireless systems operate on can not be chosen at random. Fortunately, frequency coordination with Shure PSM systems is a relatively painless process. The frequencies Shure has selected for each system are already coordinated. With PSM 600, there are 10 total frequencies available, all of which are compatible. PSM 400 has 16 frequencies, of which any eight are compatible. Lastly, PSM 700 has 32 frequencies, divided into two groups of 16 compatible – as long as all the frequencies are chosen from within the same group, they are compatible.

When combining different PSM systems, frequency coordination remains straightforward. The 16 frequencies that comprise the PSM 400 frequency set are identical to those of group 1 of PSM 700. It is important to note that when combining PSM 400 and 700, you are now limited to a maximum of eight compatible frequencies. PSM 600 operates in a different frequency range from that of PSM 400 or 700. For the maximum number of simultaneous monitor mixes, all 10 PSM 600 frequencies can be used in conjunction with 16 PSM 700 systems, for a grand total of 26 stereo mixes (whew!).

When choosing a wireless system, be it a microphone or personal monitors, there are three questions that need to be asked:

1. Where will the wireless system be used?
2. How many wireless systems will be used?

3. Are there any other existing wireless systems (microphones included) currently in use?

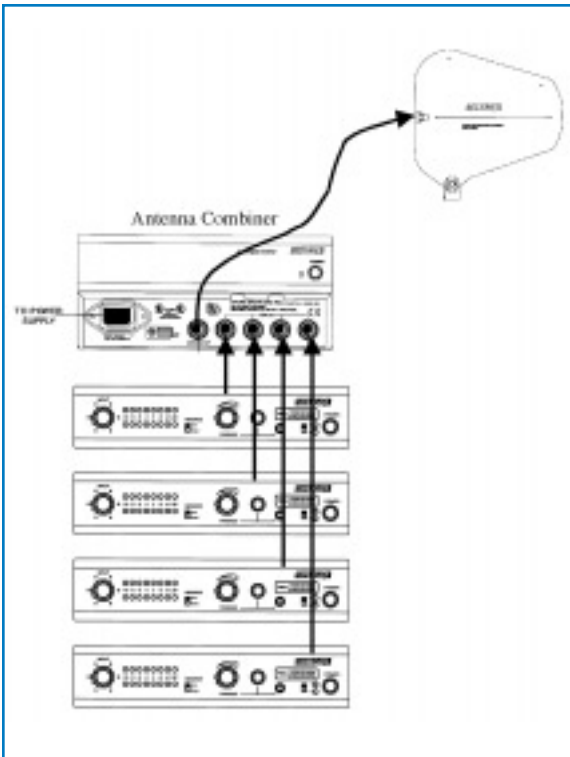
Location is critical, and can even influence the buying decision. Since wireless systems operate on unoccupied television channels, active television channels in the city where the wireless systems will be used influence the maximum number of compatible frequencies. For example, PSM 600 frequencies are spread over 6 US television channels. If three of those channels are occupied, the total number of compatible frequencies has been reduced from ten to four. In this instance, PSM 400 or 700 is a better choice if more than four total frequencies are required. For those performers who travel, it is imperative that someone be responsible for frequency coordination from city to city. Shure Applications Engineering can assist with information on local television in most major areas. The Frequency Locator Mode of PSM 700 is also a useful tool for traveling groups.

For the most part, Shure PSM and wireless microphone systems do not overlap frequency ranges in the US. Integrating a wireless personal monitor rig with wireless microphones should not require any special considerations. Be aware, though, that when using other brands of wireless, or Shure systems in frequency ranges for countries besides the US, they may operate in the same frequency range as PSM systems. Care should be taken to pick frequencies that do not conflict with wireless microphones.

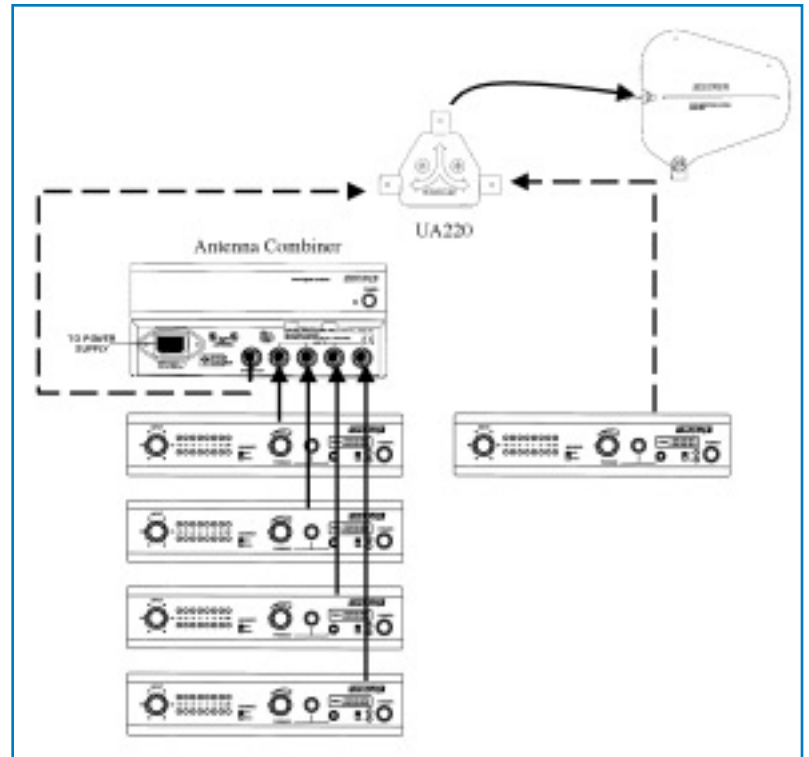
Antenna Combining for PSM 600 and 700

Antenna combining is crucial to obtaining optimal performance from Shure PSM systems. Reducing the number of transmitter antennas reduces the chance of experiencing multipath dropouts. The PA760 and PA770 Active Antenna Combiners can be used to connect up to four PSM transmitters to one antenna. For more than four systems, a passive combiner (such as the Shure UA220) can be used to connect the outputs of two active combiners. The recommended antenna to use on the output of the antenna combiner is the PA705 Directional Antenna. A half wave antenna may be substituted for the directional antenna. The 1/4 wave antenna supplied with the PSM transmitters can be used if it is attached directly to the output connector of the PA760 (or PA770), but NOT for remote antenna applications.

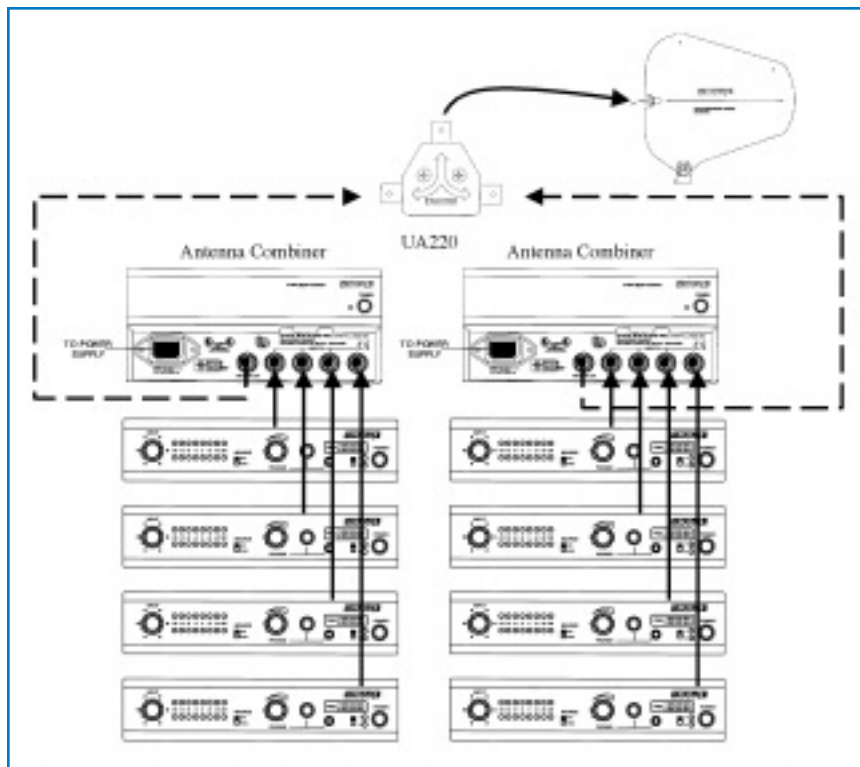
For systems involving more than eight transmitters, two entirely separate antenna combining systems should be used. Never use an active combiner on the antenna outputs of other active combiners. Additionally, do not passively combine (such as with the UA220) more than two active combiners.



Antenna Combining – 4 systems



Antenna Combining – 5 systems



Antenna Combining – 6 or more systems

Ambience

Room acoustics or natural reverberation.

Amplitude

Magnitude of strength of signal or wave.

Antenna

Electrical circuit element that transmits or receives radio waves.

Auxiliary (Aux) Send

An extra output from a mixer channel with separate level control. Usually used to create monitor mixes or as effects sends. See also: Prefader; Postfader.

Balanced

A circuit that carries information by means of two equal but opposite polarity signals, on two conductors.

Bodypack

A receiver style which can be worn on the body.

Cardioid Microphone

A unidirectional microphone with moderately wide front pickup (131 deg.). Angle of best rejection is 180 deg. from the front of the microphone, that is, directly at the rear.

Comb Filtering

The variations in frequency response caused when a single sound source travels multiple paths to the listener's ear, causing a "hollow" sound quality. The resultant frequency response graph resembles a comb.

Compressor

A signal processor that reduces the level of incoming audio signals as they exceed a given threshold. The amount of reduction is usually defined by the user.

Equalizer

A signal processor that allows the user to boost or cut selected frequencies. Used for tone shaping and limited feedback control. Variations include graphic or parametric.

Feedback

In a PA system consisting of a microphone, amplifier, and loudspeaker, feedback is the ringing or howling sound caused by the amplified sound from the loudspeaker entering the microphone and being re-amplified.

Fidelity

A subjective term that refers to perceived sound quality.

Frequency Agile

Having the ability to change frequency: tuneable.

Frequency Response

Variation in amplitude of a signal over a range of frequencies. A frequency response graph is a plot of electrical output (in decibels) vs. frequency (in Hertz).

Front-of-House (FOH)

Usually refers to the mix created for the audience, or "house."

Gain-Before-Feedback

The amount of gain that can be achieved in a sound system before feedback or ringing occurs.

Gate (Noise Gate)

A signal processor that mutes the audio signal when it drops below a given threshold.

Ground Plane

Electrical approximation of a zero-potential reflective surface at the base of an antenna.

Insert

A routing point on a mixer that allows an audio signal to be sent to an external device and returned back to the same mixer channel.

Isolation

Freedom from leakage; ability to reject unwanted sounds.

Limiter

A signal processor that prevents signal levels from exceeding a certain threshold.

Mix

A combination of input signals that are varied in level and tone to provide the desired sound for the listener.

Mixer

A device which allows the combination, manipulation, and routing of various audio input signals.

Mono

A single channel of audio.

NOM

Number of open microphones in a sound system.
Decreases gain-before-feedback by 3 dB every time the number of open microphones doubles.

Omnidirectional Microphone

A microphone that picks up sound equally well from all directions.

Operating Frequency

The final output frequency of a transmitter or the tuned frequency of a receiver.

PA

Public Address. Usually refers to a sound system.

PAG

Potential Acoustic Gain is the calculated gain that a sound system can achieve at or just below the point of feedback.

Pan

The ability to move a signal across the stereo field, from left to right, or vice versa.

Prefader

Auxiliary sends whose output levels are unaffected by the main mixer channel level control.

Postfader

Auxiliary sends whose output levels are directly affected by the output level of the main mixer channel level control.

RF

Radio frequency.

Receiver

Device which is sensitive to radio signals and recovers information from them.

Reverberation

The reflection of a sound a sufficient number of times that it becomes non-directional and persists for some time after the source has stopped. The amount of reverberation depends on the relative amount of sound reflection and absorption in the room.

SPL

Sound Pressure Level is the loudness of sound relative to a reference level of 0.0002 microbars. Usually expressed in dB SPL.

Sound Reinforcement

Amplification of live sound sources.

Stereo

Two channels of audio, left and right, which can be used for panning audio signals to simulate a realistic listening environment.

Subgroups

A way of combining audio signals from individual mixer input channels into smaller groups.

Supercardioid Microphone

A unidirectional microphone with tighter front pickup angle (115 deg.) than a cardioid, but with some rear pickup. Angle of best rejection is 126 deg. from the front of the microphone.

Transducer

A device that converts one form of energy to another. A loudspeaker transducer converts electrical energy (the audio signal) into acoustical energy (sound).

Transmitter

Device which converts information to a radio signal.

UHF

Ultra High Frequency (about 300-1000 MHz).

VHF

Very High Frequency (about 30-300 MHz).

Gino Sigismondi, a Chicago native and Shure Associate since 1997, has been active in the music and audio industry for nearly ten years. In addition to his work as a live sound and recording engineer, Gino's experience also includes performing and composing. Gino earned his BS degree in Music Business from Elmhurst College, where he was a member of the Jazz Band, as both guitar player and sound technician. After graduation, he spent several years working for Chicago area sound companies and night clubs, before settling down to a select group of the area's top local acts. As a member of Applications Engineering, Gino brings his years of practical experience to the product training seminars he conducts for Shure customers, dealers, distribution centers, and internal staff. He has also authored several Shure applications bulletins as well as written for the Shure Web site. Gino continues to remain active as a sound engineer, expanding his horizons beyond live music to include sound design for modern dance and church sound.

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ADDITIONAL SHURE PUBLICATIONS AVAILABLE:

- Introduction to Wireless Systems
- Introduction to Personal Monitor Systems
- Selection and Operation of Wireless Microphone Systems
- Shure's Microphone Techniques for Music
- Shure's Microphone Techniques for Studio Recording
- Shure/Tascam Guide to Microphones and Multitrack Recorders

These educational publications are available free of charge, as are brochures and catalogs on our full line of sound reinforcement and recording products. To request your complimentary copies, please contact us.



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